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EVALUATION OF SPEECH PROCESSING SYSTEMS

Evaluation of Electronic/Active Hearing Protectors for Use in Underground Coal Mines

Prepared for

UNITED STATES DEPARTMENT OF THE INTERIOR
BUREAU OF MINES

by

ENVIRONMENTAL ACOUSTICS LABORATORY
THE PENNSYLVANIA STATE UNIVERSITY

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This report presents a discussion of performance parameters for electronic or active hearing protectors that will meet the needs of underground coal miners. This information is incorporated into a proposed set of specifications that can be used to judge the suitability of a particular electronic hearing protector for the coal mine application. Acoustic and electrical tests were performed on a developmental model of an electronic hearing protector built by the Federal Bureau of Mines. A commercially active hearing protector, the British-built ACOS A-9000/2, was also examined. The recommended test of intelligibility for electronic hearing protectors and other communication systems as used in a coal mine setting is the Modified Rhyme Test (MRT) developed in 1965. The suitability and method of administering this test are discussed.

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DEPARTMENT OF THE INTERIOR
BUREAU OF MINES
WASHINGTON, D.C.

Disclaimer Notice

FOREWORD

This report was prepared by The Environmental Acoustics Laboratory of the Pennsylvania State University under USBM Grant No. G 0155032. The grant was initiated under the Coal Mine Health and Safety Research Program. It was administered under the technical direction of the Pittsburgh Mining and Safety Research Center with Mr. John Durkin 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 January 1975 to 30 April 1976. This report was submitted by the authors on 3 May 1976.

This technical report has been reviewed and approved.

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ABSTRACT

This report presents a discussion of performance parameters for electronic, or active, hearing protectors that will meet the needs of underground coal miners. This information is incorporated into a proposed set of specifications that can be used to judge the suitability of a particular electronic hearing protector for the coal mine application. Acoustic and electrical tests were performed on a developmental model of an electronic hearing protector built by the U. S. Bureau of Mines. This model showed only a slight reduction in speech intelligibility which was traced to poorly matched components and in particular to the transducers. Tests of alternative components showed the Knowles BL-1671 microphone to be suitable for this application because of its performance and extremely small size. The Roanwell H-143/AIC earphone was found to have a good frequency response but has the disadvantage of being rather bulky which causes a slight degradation of the low frequency attenuation of the hearing protector. A satisfactory installation of these components in a hearing protector muff was demonstrated and tested. Results of tests of several methods of controlling the sound level under the hearing protector muff showed that a simple diode clipper circuit can be used with only a negligible loss in speech intelligibility for normal-hearing listeners. A commercially available active hearing protector, the British-built ACOS A-9000/2 was examined. This hearing protector provides sound level limiting through a saturating amplifier. Preliminary tests have shown that the performance characteristics of this commercial unit meet many of the proposed specification, so further testing and evaluation should be performed to determine its suitability for use in underground coal mines. The recommended test of intelligibility for electronic hearing protectors and other communication systems as used in a coal mine setting is the Modified Rhyme Test (MRT) developed by House et al. in 1965. The suitability and method of administering this test are discussed.

Table of Contents

Page

Disclaimer Notice i
Forward ii
Project Personnel iii
Abstract iv
Table of Contents v
List of Figures vii
List of Tables x
1 Development of electronic hearing protectors 1
1.1 Introduction . . . .
1.2 Transfer function of ideal electronic hearing protectors 1
1.2.1 Frequency response of passive hearing protectors. 1
1.2.2 Gain of the communication system 2
1.2.3 Frequency response of the active hearing protector system 2
1.2.4 System transfer function 3
1.2.5 Electronic transfer function 4
1.2.6 Signal compression techniques 5
1.2.7 Conclusions and specifications 6
1.3 USBM Developmental hearing protector system 7
1.3.1 Tests performed 7
1.3.2 Phase response 8
1.3.3 Response of USBM system in diffuse field 9
1.3.4 Development of observed responses from component specifications 9
1.3.5 Measured response of USBM system components 10
1.3.6 Conclusions 10
1.3.7 Tests of alternative components 11
1.3.8 Conclusions 13
1.4 Intelligibility tests of some output limiting methods . . 13
1.4.1 Introduction to Experiment I 13
1.4.2 Subjects 13
1.4.3 Stimuli 15
1.4.4 Apparatus and procedure 15
1.4.5 Results 17
1.4.6 Conclusions from Experiment I 17
1.4.7 Introduction to Experiment II 19
1.4.8 Stimuli 19
1.4.9 Apparatus and procedure 19
1.4.10 Results 19
1.4.11 Conclusions from Experiment II 19
2 Tests of ACOS A 9000/2 Hearing Protector 23
2.1 Introduction 23
2.2 Tests of the hearing protector system 23
2.3 Response of electronic system components 24
2.4 Hearing protection of the ACOS muff 25
2.5 Conclusions 25
3 Evaluation of ACOS A 9173 Hearing Protector 27
3.1 Attenuation Performance 27
3.2 Speech Discrimination Performance 29

		Page
3.2.1	Subjects	29
3.2.2	Equipment	29
3.2.3	Test Material	29
3.2.4	Test Muffs	30
3.2.5	Procedures	30
3.2.6	Results	30
3.3	Conclusions	34
4	Testing Speech Intelligibility of Communications Systems for Use in Coal Mine Environments	37
4.1	Introduction	37
4.2	The Phenomenon of Speech Intelligibility	37
4.3	Intelligibility Testing	48
5	Modifications to the EAL-USBM Training Course	52
5.1	Training Course in Hearing Conservation	52
5.2	Automation of Hearing Conservation Training Course	52
5.2.1	Slide Synchronization Systems	52
5.2.2	Choice of Synchronization System for USBM	53
5.2.3	Magnetic Tape Format	53
5.2.4	Method of Presentation	54
6	References	57
7	Appendix A	132
8	Appendix B	136

List of Figures

Figure		Page
1	Mean attenuation of five different types of hearing protectors	62
2	Recommended frequency response for electronic hearing protector communication system	62
3	Electronic hearing protector response for some representative ambient noise levels	63
4	Transfer function of an ideal electronic hearing protector	64
5	Electronic transfer function reference grid	65
6	Block diagram of electronic ear protector transfer function simulator	66
7	A-Single-stage diode clipper	67
	B-Isolated two-stage diode clipper	67
	C-Non-isolated two-stage diode clipper	67
8	Transfer characteristics of diode clippers	68
9	Transfer function of dbx compressor	69
10	Equipment arrangement for measuring pure-tone response of hearing protectors in diffuse field room	70
11	Sound field 100 dB	71
12	Electronic muff USBM No 1, Sound field 100 dB, Electronics off	71
13	USBM Electronic muff No 1, Sound field 100 dB	72
14	USBM Electronic muff No 1, Sound field 90 dB	72
15	USBM Electronic muff No 1, Sound field 80 dB	73
16	USBM Electronic muff No 1, Sound field 70 dB	73
17	Response of USBM electronic hearing protector for representative noise levels	74
18	Contribution of electronic system response to muff attenuation	75
19	Electronic muff USBM No 2, Sound field 100 dB, electronics off	76
20	Electronic muff USBM No 2, Sound field 100 dB, left side	76
21	Electronic muff USBM No 2, Sound field 100 dB, right side	77
22	Electronic muff USBM No 6, Sound field 100 dB, left side	77
23	Electronic muff USBM No 2, Sound field 100 dB, hearing aid test box	78
24	Electronic muff USBM No 2, Sound field 90 dB, hearing aid test box	79
25	Electronic muff USBM No 2, Sound field 80 dB, hearing aid test box	80
26	Attenuation of MSA electronic system in sound fields of 80 dB, 90 dB, and 100 dB Lp pure tone	81
27	Electronic muff USBM No 2, 100 dB diffuse field, electronics off	82
28	Electronic muff USBM No 2, 115 dB diffuse field, electronics on	82

Figure		Page

29 Electronic muff USBM No 2, 110 dB diffuse field, electronics on . . . 83
30 Electronic muff USBM No 2, 100 dB diffuse field, electronics on . . . 83
31 Electronic muff USBM No 2, 90 dB diffuse field, electronics on . . . 84
32 Electronic muff USBM No 2, 80 dB diffuse field, electronics on . . . 84
33 Electronic muff USBM No 2, 70 dB diffuse field, electronics on . . . 85
34 Sound pressure level under MSA Comfo 600 protector . . . 86
35 Attenuation of MSA Comfo 600 protector with electronic gain control . . . 87
36 Transfer characteristic of USBM-MSA electronic protector No 6 . . . 88
37 Manufacturer's response curves and resulting composite response . . . 89
38 Sound field, pink noise . . . 89
39 USBM D-371 without 2 K Ohm resistor . . . 90
40 USBM D-371 with 2K Ohm resistor . . . 90
41 $4 Lapel crystal microphone . . . 91
42 $10 Dynamic Hi. Z. microphone . . . 91
43 D-526 Earphone in 6 cc coupler . . . 92
44 D-426 Earphone; Comparison between measured and published response curves . . . 92
45 D-426 Earphone in MSA 600 muff on flat plate coupler . . . 93
46 Composite response of USBM system from response of components . . . 93
47 Manufacturer's specifications for sensitivity of BL-1671 mic. . . . 94
48 Sound field 74 dB/1/3 octave . . . 94
49 BL-1671 Mic in MSA 600 with 1" tube; no cotton . . . 95
50 BL-1671 Mic in MSA 600 with 1" tube; cotton ball in tube. 95
51 Earphone H-143/AIC on 6 cc coupler . . . 96
52 Earphone H-143/AIC in MSA 600 protector on flat plate . . . 96
53 Earphone H-143/AIC in MSA 600 protector with rubber baffle mounting . . . 97
54 Earphone H-143/AIC in MSA 600 protector with rubber baffle sealed with tape . . . 97
55 Earphone mounted on 3/8" closed cell neoprene . . . 98
56 Earphone and components mounted in open cell foam . . . 98
57 Earphone H-143/AIC mounted on baffle in MSA 600 . . . 99
58 Earphone H-143/AIC mounted in open cell foam . . . 99
59 Real-ear attenuation of hearing protector mock-up based on MSA Comfo 600 hearing protector . . . 100
60 Signal compression networks . . . 101
61 Test circuit for compression networks . . . 102
62 Response of speech filter . . . 103
63 ACOS A 9000 Electronic muff; electronics off . . . 104
64 ACOS A 9000 Electronic muff; electronics off; 80 dB sound field . . . 104
65 ACOS A 9000 Electronic muff; electronics on; 115 dB sound field . . . 105

Figure		Page

66 ACOS A 9000 Electronic muff; electronics on; 110 dB sound field 105
67 ACOS A 9000 Electronic muff; electronics on; 100 dB sound field 106
68 ACOS A 9000 Electronic muff; electronics on; 90 dB sound field 106
69 ACOS A 9000 Electronic muff; electronics on; 80 dB sound field 107
70 ACOS A 9000 Electronic muff; electronics on; 70 dB sound field 107
71 Attenuation of ACOS A 9000 protector 108
72 R & L ACOS A 9000 Attenuation for 100 dB input 109
73 R & L ACOS A 9000 Attenuation for 90 dB input 110
74 R & L ACOS A 9000 Attenuation for 80 dB input 111
75 R & L ACOS A 9000 Attenuation for 70 dB input 112
76 R & L ACOS A 9000 Attenuation for 60 dB input 113
77 Attenuation of the electronics of the ACOS A 9000 protector 114
78 ACOS A 9000 protector earphone only in muff on flat plate 114
79 Sound field 74 dB/1/3 octave, ACOS mic. 115
80 ACOS system mic. and earphone 115
81 ACOS A 9000/2 circuit diagram showing added input and output connections 116
82 (A through E) Waveforms at pin 3 of ACOS integrated circuit 117
83 Transfer charateristics of ACOS A 9000 hearing protector. 119
84 Frequency response of ACOS A 9000 electronic hearing protector with electrical input 120
85 Acoustic output of ACOS hearing protector for pink noise input 121
86 Comparison between real-ear attenuation of ACOS A 9000/2 hearing protector and published specifications 122
87 Performance of ACOS A 9000 electronic hearing protector . 123
88 Sound field 124
89 Attenuation of ACOS A 9173 in 70 dB/1/3 octave sound field; electronics off 124
90 Attenuation of ACOS A 9173 in 60 dB/1/3 octave sound field; electronics on 125
91 Attenuation of ACOS A 9173 in 70 dB/1/3 octave sound field; electronics on 125
92 Attenuation of ACOS A 9173 in 80 dB/1/3 octave sound field; electronics on 126
93 Attenuation of ACOS A 9173 in 90 dB/1/3 octave sound field; electronics on 126
94 Electronic transfer function for ACOS A 9173 peak limiting ear defender 127
95 Block diagram of test facility for evaluation of hearing protectors.. 128
96 Two-Track Stereo head configuration for 1/4" tape 129
97 Four-Track Stereo head configuration for 1/4" tape . . . 129
98 Two-channel configuration for 3M cassette system 130
99 Stereo track configuration for cassette tapes 130
100 Equipment connections for automated slide presentation . 131
A1 Spectra of Underground Coal Mining Machines 133
A2 Weighted Average Maximum Noise Spectrum for Underground Coal Mines 135

List of Tables

		Page
I.	Real-ear attenuation measurements	14
II.	System test conditions	16
III.	Listener test scores	18
IV.	Comparison of test scores	20
V.	Mean Test scores	21
VI.	ACOS (A 9000) Active Muff: Anechoic test	26
VII.	Attenuation of ACOS A 9173 protectors	28
VIII.	Signal levels used for measurement of speech discrimination with hearing protectors	31
IX.	Analysis of variancd	32
X.	Mean discrimination scores averaged across protector condition	33
XI.	Mean discrimination scores for protector condition averaged across listening condition	35
XII.	Mean discrimination scores for individual protector and listening condition combinations	36
XIII.	Slide sequence for automated training program	55
A1.	Calculation of weighted average maximum Lp from octave band spectra	134

1. DEVELOPMENT OF ELECTRONIC HEARING PROTECTORS

1.1 Introduction: Noise exposure of underground coal miners involves periods of high level noise when hearing protection would be required to prevent noise-induced hearing loss, and periods of quiet when hearing protectors are not needed and they might be a detriment to the detection of roof talk or other warning signals. Consequently, a hearing protector for use in underground coal mines should be capable of fast and convenient removal and replacement, or be capable of turning itself "off" or "on" depending upon the noise level in the immediate area. This requirement has led to the development of an electronic hearing protector that provides automatic on-off operation.

1.2 Transfer Function of Ideal Electronic Hearing Protector: The purpose of an electronic hearing protector for use in underground coal mines is to provide an electronic communication system which allows speech and warning signals to be heard in as natural a way as possible while hearing protectors are being worn. The electronic system would include a microphone, an earphone, and a level control circuit designed to limit the level of signal passed through the muff so that the signal level under the ear protector muff does not exceed a pre-set sound pressure level. The design limit presently proposed is 90 dB(A). For sound levels above the design limit, the electronic communication system should be essentially turned off so that it does not contribute to the sound level at the protected ear. In this way, the protector will allow normal communication of voice and roof warning signals while machinery is quiet and still provide protection of hearing when noisy machinery is operating. This kind of protector would be particularly valuable in many mining work areas because the duty cycle of mining machinery is such that it is not practical to remove the protectors every time the machines are shut down.

In the following discussion, the term "System transfer function" refers to the relation between the sound level under the electronic hearing protector muff (with electronics on) and the ambient sound impinging on the complete electronic hearing protector system. The sound level observed under the muff is made up of sound reaching the ear through the hearing protector muff combined with sound reproduced by the earphone of the electronic communication system. The term "Electronic transfer function" refers to the acoustic output of the earphone of the electronic system as a function of the acoustic signal picked up by the microphone outside the hearing protector muff.

1.2.1 Frequency Response of Passive Hearing Protectors: The attenuation provided by conventional, passive, muff-type hearing protectors is frequency dependent as indicated in Fig. 1. This curve is drawn through data points representing the mean attenuation of all brands of muffs tested by Michael, Bolka (1972). The shape of this curve explains the unnaturalness of sounds heard through the protectors. All sounds heard while wearing conventional hearing protectors will be modified approximately by this response curve.

1.2.2 Gain of the Communication System: To restore natural-sounding communications, the electronic system in an active hearing protector should provide a signal input to the ear with approximately zero loss when background is quiet. The exact gain or loss of the electronic system should be based on subjective evaluation of the naturalness of the reproduced sound and will be affected by the bandwidth of the communication system and the residual background (biologic) noise under the muff. The electronic system gain must be measured acoustically. The gain of the system is defined here to be the difference in dB between sound pressure level produced by the earphone inside the muff and sound pressure level at the protector microphone. The system gain for research models should be designed for zero dB with a range of adjustment of at least +3 dB to -6 dB. When the actual gain requirement has been established, the gain can be set by fixed internal components.

1.2.3 Frequency Response of the Active Hearing Protector System: At first thought, it would appear that the frequency response should be as wide as possible for "high fidelity" reproduction to preserve naturalness and possible equalization to flatten the frequency response. However, a closer look at the problem indicates that equalization and wide frequency response are both unnecessary and undesireable at times.

Fletcher (1929) has stated that the telephone response of 300 Hz to 3000 Hz is the minimum bandwidth necessary for recognition of the talker. The response within this band should be relatively flat to preserve naturalness of the voice. Assuming that this pass band is a desirable goal for the electronic hearing protector system, the remaining characteristics can be derived.

An idealized frequency response curve is shown in Fig. 2. Persons with recruitment*may find the electronically reproduced signal unpleasant and intolerable due to the action of the automatic level control circuit unless frequencies above 3000 Hz are attenuated rapidly. A slope of at least 12 dB per octave should be acceptable for most persons with recruitment. For persons with normal hearing, the telephone quality of 3000 Hz response should also be acceptable. Since the attenuation of the muff is only about 15 dB for frequencies below 300 Hz the sound passing through the muff and the sound introduced by the electronic system will be combined in this frequency range. An attenuation slope of at least 6 dB per octave below 300 Hz should be adequate to prevent unpleasant combinations of sounds and to reduce the size of components required for the electronic circuity. The idealized curve in Fig. 2 shows these frequency response design points and the recommended minimum rate of attenuation outside this pass band. The dashed lines indicate the probable real frequency response if the system is designed according to these recommendations. In reality the frequency response will probably be dominated by the mechanical cutoff frequencies of the microphone and/or earphone and, providing the overall response does not contain extreme peaks or dips, the system should perform adequately.

* See Appendix

It should be kept in mind that at low ambient noise levels the electronic system response (Fig. 2) will dominate. At high noise levels where the automatic level control has turned off the electronics, the muff response (Fig. 1) will dominate. Figure 3 shows the predicted response of such an electronic hearing protector at a few representative noise levels. These curves were produced by combining the curves of Fig. 1 and 2 at the indicated noise levels. It is obvious from these curves that equalization to compensate the sloping response of the ear muff is not practical.

1.2.4 System Transfer Function: It it is assumed that the automatic level control circuitry should prevent the sound level under the muff from exceeding 90 dB(A) and that a through-the-muff gain of 0 dB is desired when the ambient noise level is below 90 dB(A) the characteristic curve for the gain control circuit can be derived.

First, the R-factor (Giardino, USDI 1975) for the hearing protector to be used must be determined. For the average muff of Fig. 1, this is 26 dB. If the attenuation of the muff is reduced by two standard deviations (approximately 10 dB) to allow protection for 90% of the wearers, the R-factor becomes 16 dB. This means that a sound level of 100 dB(A) would be attenuated by the hearing protectors to 74 dB(A) or 84 dB(A) depending upon which R-factor is used.

The sound level under the muff is determined from the ambient noise level and the R-factor of the muff combined acoustically with the signal introduced under the muff by the electronic system. This sound level should not exceed the exposure level design limit, for example, 90 dB(A).

The required system transfer function can then be derived as follows: Assume that the limiting exposure level under the muff is to be 90 dB(A) in accordance with USDI recommendations (USDI Federal Register 1974). Referring to Fig. 4 the ideal electronic hearing protector system should transfer acoustic signals, (i.e. communications, warning signals, and noise) linearly at 0 dB gain for sound levels up to an ambient level of 90 dB(A). This performance is represented as the solid line sloping upward at a 45 degree angle to 90 dB(A) in the lower left hand region of Fig. 4. This part of the system transfer function is designated as Linear Region I. As the ambient sound level increases above 90 dB(A) the electronic system should act to reduce the level of sound transmitted through the hearing protectors so that the combined level of the electronically introduced sound and the muff attenuated sound does not exceed 90 dB(A). Therefore, the system transfer function should become horizontal at the 90 dB(A) level.

It would be desirable for the sound level under the muff to remain at 90 dB(A) for all higher ambient sound levels. Unfortunately, the sound level under the muff will not remain at the limiting level as the ambient noise level continues to increase because the sound attenuation provided by the ear protector is limited. At some higher ambient level the sound passing through the hearing protector muff will become dominant and the sound level under the protector will follow the line marked LINEAR REGION II. The LINEAR REGION II line is parallel to the LINEAR REGION I line but is displaced vertically by the R-factor of the protector. In this example the vertical displacement corresponds to an R-factor of 26 dB. The ideal system transfer function of an electronic hearing protector based on a muff with an R-factor of 26 dB, should then follow the solid line in Fig. 4. Unfortunately, the electronic system input complicates the actual system transfer function.

1.2.5 <u>Electronic Transfer Function</u>: If the output of the electronic system of an active hearing protector is designed to remain constant as shown in Fig. 4, the overall sound level presented to the ear will follow the solid line to 106 dB(A) input level and the dotted line from 106 dB(A) input level to 120 dB input level where it joins the Linear Region II line. The sound level departs from the ideal solid line response because the acoustic energy transferred through the protector becomes significant above 106 dB(A) in this example. Therefore, to maintain the 90 dB(A) limit for higher ambient noise levels, the electronic transfer function should follow the dashed curve in Fig. 4. If the electronic transfer function follows this dashed curve, the system transfer function for the example (protector R-factor of 26 dB) will follow the solid line between Linear Regions I and II and the 90 dB(A) limit will be held until the ambient noise level reaches about 116 dB(A).

The design goal for the electronic transfer function has been replotted in Fig. 5. This curve will be used for reference in plotting transfer functions of electronic hearing protectors and components in the remainder of this report.

A set of equipment was assembled to simulate the system transfer function of an electronic hearing protector as described in Fig. 4. The equipment was connected as shown in the block diagram Fig. 6. Physically, the microphone was mounted on the headphone band to be as close as possible to the ear as it would be in an electronic hearing protector system.

Referring to Fig. 6, the signal from the microphone was amplified by an Altec 1591A compressor amplifier. The function of this compressor is to limit the level of the signal at its output by providing an automatic-gain-control function. The Cinema equalizer provides a variable slope to a system pass-band to compensate for non-uniform frequency response of other system components such as the hearing protector muff. The Krohn-Hite 3750 filter provided a variable high- and low-pass filter function to adjust the bandwidth of the overall system. The attenuator allowed control of the system gain and the final amplifier provided an impedance match to the 8-Ohm earphones.

Although no extensive listener tests were performed with this system, some valuable conclusions were drawn. The system produced a very clean and natural sound even with the filters set to pass only 300 Hz to 3000 Hz with 24 dB/octave out-of-band attenuation. Within the pass band the effects of equalization within the capabilities of the Cinema 4031B could not be discerned. It was apparent that much more level compression would be required for an electronic hearing protector than could be realized with the Altec 1591A volume compressor.

The extreme level compression called for by the curve of Fig. 4 could be obtained by using a dbx 119 dynamic range enhancer (volume level compressor-expander). Although this unit is fairly sophisticated and too expensive to be considered for use in the electronic hearing protector, it provided an opportunity to test a system having the required compression characteristics.

1.2.6 Signal Compression Techniques: The performance curves shown in Fig. 5 for Electronic Hearing Protectors indicate that the amplification must cut off at a fairly rapid rate above the limiting level. Performance matching this curve may require rather sophisticated electronic circuitry. Optimum performance requires that electronic system output should tend to decrease as sound level increases at very high levels.

As a first step to obtain the desired gain characteristics, a simple diode clipper was constructed and tested. The circuit is shown in Fig. 7A with output at point A. (The second clipper stage was not present during tests of the single-stage clipper.) The operational amplifiers are each connected as voltage followers to provide a very low source impedance for driving the clipper circuit and a very high input impedance to minimize loading effects.

The transfer characteristic of the single-stage diode clipper is shown in Fig. 8. In the circuit (Fig. 7A), R = 330 ohms and D = 1N 456. A closer approximation to the required curve was obtained by adding a second clipper stage. The transfer characteristic of the two-stage clipper is also shown in Fig. 8. In the two-stage circuit, the diodes were type 1N5282 with R = 330 ohms. The circuit is shown in Fig. 7B with output taken at point B. Adding a third diode clipper stage resulted in essentially no change from the two-stage curve.

When the two clipper stages were coupled without isolation as in Fig. 7C, only a slight rounding of the characteristic curve resulted. Increasing the resistor R to 18 K ohms degrades the transfer characteristic curve and is not recommended.

An almost perfect match to the required characteristic curve was obtained with a dbx 119 dynamic range enhancer (signal level compressor) set for infinite compression. By suitable adjustment of the compression threshold, the curve in Figure 9 was obtained. The dbx 119 employs fairly sophisticated circuitry and if it could be incorporated into an electronic hearing protector it would no doubt result in a prohibitively expensive unit. However, the distortion introduced by the dbx 119 is negligible and the input-output characteristic represents a nearly ideal design goal.

1.2.7 Conclusions and Specifications: An electronic hearing protector appears to be a satisfactory alternative to conventional passive ear protectors that must be frequently removed and replaced in an underground coal mine environment. Based on performance considerations discussed in Section 1.2 of this report and other obvious requirements, a set of specifications is proposed for an electronic hearing protector for use in underground coal mines.

These design specifications have been based on the mean attenuation performance of the basic hearing protector. It should be realized that 50% of the population will receive less protection than the mean value indicates and adjustments should be made to provide the proper confidence limits for any hearing conservation program.

SPECIFICATIONS FOR ELECTRONIC HEARING PROTECTOR

Attenuation of Basic Muff: R-factor of 25 dB or greater

Maximum Sound Level under Muff: Determined by current USBM regulations

Ambient levels up to 180 dB should not impair subsequent operation of the hearing protector system.

Combined Electrical and Acoustic Transfer Characteristics of System: System transfer characteristics should follow curve of Fig. 5 up to 115 dB(A) for optimum performance.

Electronic System Transfer Characteristics

Gain = 0 dB ±2 dB at 2000 Hz, 80 dB Lp
Frequency Response ±6 dB between 300 Hz and 3000 Hz
6 dB/octave below 300 Hz
-12 dB/octave above 3000 Hz
Distortion : Less than 3% THD at 80 dB Lp
Directivity: Not critical--Right and left sides should be independent. Microphone directivity not essential.
Battery Life (if separately battery powered) Minimum 50 hours
Electronic system should be of Fail-Safe design such that catastrophic failure of any of the electronic components will not allow electronic signal transfer to exceed 90 dB(A).

Mechanical: Electronic Hearing Protector including Safety Helmet (if attached) should function satisfactorily and safely after 6-foot drop to a concrete floor.
System should function in Relative Humidity up to 100%
Hearing protector should be comfortable to wear for extended periods.
Seals should be easily replacable

1.3 USBM Developmental Hearing Protector System: The first models of the USBM electronic hearing protector were tested by EAL using speech and roof talk discrimination tests with human subjects. The results of these tests were reported by Michael, et.al. (1973). Although the performance of the system was quite good, it was concluded that there was considerable room for improvement and that additional tests and development were needed.

One of the results of these tests indicated that intelligibility was slightly degraded by the electronic system at and below noise levels of 80 dB Lp. While discrimination scores of normal-hearing persons using the electronic protector were all above 90%, (a value considered normal) the test scores for low noise levels where the wearer is dependent upon communication through the electronic system showed significant differences indicating that the electronic system was degrading intelligibility. The speech discrimination tests were unable to show any differences in performance when damping resistors were used to smooth out the frequency response of the microphone; consequently, these damping resistors were eliminated. Variations in transfer gain also showed only minor differences, although it was concluded that the electronic system should result in an acoustic transfer gain of 0 dB at low noise levels. Differences in speech discrimination abilities were observed among the eight units but were not correlated to specific units.

1.3.1 Tests Performed: Differences between models of the original electronic muff system could not be determined since many of the units have been modified or are no longer available. However, two of the original systems using A-0 muffs and two of the systems built with MSA Comfo 600 muffs were made available to EAL for testing. The ear protector systems were tested on an artificial test head (Michael and Bolka, 1972) in the center of the diffuse field test room as shown in Fig. 10. The compressor microphone is a B & K condenser microphone placed as close as possible to the electronic hearing protector microphone to control the level of the sound field through the compressor circuit in a B & K 1022 oscillator. Although this compressor regulator system is designed for use in a non-reflective room, the regulator helps to smooth out large standing wave peaks in the diffuse field room.

With the ear protector removed, the sound field at the microphone (ear) of the artificial head is shown in Fig. 11. Although the high and low frequency response is limited by the loudspeakers and there are many standing waves present, the curve is adequate to describe the sound field.

Figure 12 shows the signal received by the artificial head with the USBM electronic muff in place but with the electronics turned off. The sound field was set at 100 dB at 1000 Hz. Care was taken to insure that the muff was sealed against the artificial skin.

The response of the electronic muff system for various sound pressure levels with the electronics turned on may be seen in Figures 13 through 16.

The sound field was set at 100 dB re .0002 dynes/cm^2 at 1000 Hz for the curves of Figures 11, 12, and 13. Figures 14, 15, and 16 show the electronic muff system response for sound fields of 90 dB and 70 dB respectively. (Note that the vertical scale has been changed in Figures 15 and 16.)

For comparison, all of the frequency response curves from Figures 13 through 16 are shown in Fig. 17 with the sound field for reference (upper curve). Level variations due to standing waves in the room have been partically smoothed out in copying.

The frequency range of influence of the electronic system at 100 dB Lp is shown in Fig. 18. The upper curve is the sound field reproduced again for reference. With the electronic system off (lower curve), the muff acts as a low pass filter with upper cut-off frequency at about 400 Hz. Communication information which occupies the spectrum above this frequency is greatly attenuated and results in an unnatural sound with reduced intelligibility. With the electronics turned on (middle curve), the system frequency response is extended upward to 3000 Hz. This frequency range is generally considered adequate for speech communication and is the band width used for telephone systems.

In this particular case, the very strong peak in response at 2300 Hz gives the electronic sound a "crisp" quality, which, in a moderate amount, should not ordinarily be objectionable and can tend to improve intelligibility. However, this particular peak is 20 dB above the other frequencies passed by the electronic system and, for practical purposes, results in an effective response only over the frequency range from 2000 Hz to 3000 Hz. The dominance of this narrow band should account for the reduction in speech discrimination scores observed during the 1973 tests by Michael, et. al.

1.3.2 <u>Phase Response</u>: The phase of the acoustic signal arriving at the reference microphone was compared with the phase of the signal arriving at the microphone in the artificial head by forming a Lissajous pattern on an oscilloscope. The Lissajous figure was observed as the signal was scanned through the frequency range.

The phase between the sound outside the muff and the sound inside the muff passed through 360° shifts continuously and rapidly as the frequency was scanned. In fact, the phase fluctuated so rapidly that it was extremely difficult to observe the phase changes. Because of the extremely rapid variations in phase, it was impossible to determine whether the electronic system resulted in a net phase reversal. It seems logical to conclude from this that phase distortion in this case is not the dominant factor affecting intelligibility and that the speech discrimination scores obtained in the tests described are primarily influenced by the frequency response of the system.

Figures 19, 20, 21, and 22 show the frequency response of USBM electronic ear protector numbers 2 and 6. The sound field at the compressor microphone was set for 100 dB at 1000 Hz. The signal picked up under the muff by the test head is reproduced in Fig. 19 for reference. Note that the 2300 Hz peak is present in each of the curves in Figures 20 through 22. (The right side of muff No. 6 was inoperative at the time of these tests.)

To verify that this 2300 Hz peak was in the electronic system and not at artifact of the test set-up, the Muff No. 2 was tested in the same manner as a hearing aid in the B & K Hearing Aid Test Box. The muff was placed on a flat plate coupler and the microphone was placed inside the test box. Figures 23, 24, and 25 show the response of the system for 100 dB, 90 dB, and 80 dB Lp input. The curve below 100 Hz is an artifact of the test set-up and should be disregarded. These three curves have been plotted on the same scale for direct comparison in Figure 26. Since the muff is not in the sound field, these curves represent the frequency response of the electronic system only.

1.3.3 Response of USBM System in Diffuse Field: To avoid the problems of standing waves resulting from the use of pure tones in the test rooms a diffuse pink noise field was set up and equalized to be constant Lp measured in one-third octave bands from 63 Hz to 8000 Hz. The electronic muff system was placed on the artificial head, and the sound under the muff was analyzed with the B & K 2113 one-third-octave spectrometer. Figure 27 shows the spectrum of sound under the muff with the electronics off. Figures 28 through 33 show the spectra observed with the electronics on for sound fields from 115 dB to 70 dB as indicated.

The spectra of Figures 27 through 33 have been copied onto the same scale in Figure 34 to allow comparison. Since each curve is the result of a different sound level, the data can be normalized to show attenuation of the muff for the various sound levels. Figure 35 shows this plot of attenuation and further illustrates the action of the automatic gain control in the electronic system.

The acoustic transfer function (Fig. 36) was measured on the artificial test head for the left side of USBM-MSA unit No. 6. The unit appears to have good level regulation although the overall level of the characteristic curve is about 6 dB high. This may be due to the sharp resonant peak shown in Figure 22. With the electronics off, the R-factor was measured to be 17 dB. The low R-factor observed here is probably caused by acoustic leaks around the cable hole in the earcup and may not be typical of this protector in general.

1.3.4 Development of Observed Response from Component Specifications: Frequency response curves supplied by Dyna Magnetic Devices Inc., for D 426 earphone and the D 371 microphone are reproduced on the standard frequency grid in Figure 37.

Assuming that the response of the electronics is flat over this range of frequencies, the two curves can be combined to obtain the composite response, also plotted in Figure 37. Because both units have peaks in response in the vicinity of 2000 Hz, the combined response shows an exaggerated peak at about 1700 Hz. Keep in mind the fact that the microphone curve shown here is stated to represent the response with a 2000 ohm damping load resistor. This resistor is not used in the present system because its effect on intelligibility was found to be negligible in the initial testing of the USBM-AO electronic muff system. The peak in response shown in Figure 26 occurs at 2300 Hz and can shift up or down in frequency because of production variations in individual units or differences in physical mounting within the ear muff cup.

1.3.5 Measured Response of USBM System Components: 1. Microphone: The diffuse sound field was again set up with the resulting spectrum shown in Figure 38. The D 371 microphone was removed from the helmet and was suspended in the center of the room. The signal was amplified by a 20 dB pre-amplifier having a 1 Megohm input impedance. The 1/3 octave spectrum of the microphone signal is shown in Figure 39. A peak in response appears in the 2000 Hz band which may contribute to the response peak of the composite system. Since the manufacturer suggested the use of 2 K ohm load resistor, the loaded response was measured and appears in Figure 40. The result is slightly flatter than the advertised response shown in Figure 37. The 2 K ohm resistor appears to improve the quality of the microphone but does not make a significant change in the overall frequency response. Response curves of two other low-cost microphones were measured in the same way and are included in Figures 41 and 42 for comparison.

2. Earphone: The frequency response of the D 426 earphone was measured by supplying the earphone with a constant voltage, pure-tone electrical signal from B & K 1022 beat frequency oscillator. The earphone was placed on a 6 cc coupler as suggested by the manufacturer. The response with .09 Vrms signal source is shown in Figure 43. This curve does not resemble that supplied by the manufacturer (See Figure 37). The specified and measured responses of the D 426 earphone are compared in Figure 44.

The response of the D 426 earphone as mounted in the MSA Comfo 600 muff on the flat plate coupler is shown in Figure 45. Combining this measured response with the measured response of the unloaded microphone from Figure 39 results in the composite response curve in Figure 46. This composite response has all the principle features of and is quite similar to the measured response of the electronic systems shown in Figure 26.

1.3.6 Conclusions: The frequency response curves in Figure 26 show why the USBM electronic hearing protector yielded reduced discrimination scores (Michael, et.al, 1973). Although the design principle is sound, the sharp peak at 2300 Hz causes frequency distortion which results in reduced intelligibility. This peak in response results from a discrepancy between the actual response and the advertised response of the D 426 earphone (Fig. 44). The measured frequency response of the D 371 microphone (Fig. 40) compares favorably with the manufacturer's specifications (Fig. 37).

The D 371 microphone would be suitable for use in an electronic hearing protector system if installed with adequate machanical protection and if the frequency response and sensitivity are not altered significantly by the mounting.

1.3.7 Tests of Alternative Components Knowles BL - 1671 Microphone: The BL-1671 hearing aid microphone is a subminiture unit designed for use in compact hearing aids such as those incorporated in eyeglass frames. The published frequency response has been re-plotted on the standard grid for reference in Fig. 47. The microphone unit is equipped with a tublar sound inlet and a built-in FET pre-amplifier. The pre-amplifier requires a 1.4 volt power supply connected through a third terminal on the microphone unit. This voltage can be easily supplied in a working system through a simple resistor-diode network and does not appear to present a particular disadvantage to the use of this microphone. The physical size and ruggedness of the microphone are distinct advantages that make the microphone an excellent choice for use in an electronic hearing protector.

The frequency response shown in Fig. 47 should be acceptable for the electronic system. However, a private communication with the designer of the microphone indicated that the response could be further improved by the process of installation. To illustrate: The microphone was placed in the diffuse sound field equalized flat as shown in Fig. 48. A one-inch long piece of hearing aid tubing was fitted to the sound inlet of the microphone resulting in a broadening of the resonant peak accompanied by a shift of this peak toward lower frequencies as shown in Fig. 49. This tubing could be used to advantage in packaging the electronic system in the hearing protector cup so that the microphone would be isolated from the shell vibration and also be protected from external dust and moisture.

Final smoothing of the frequency response was accomplished by inserting a small wad of cotton into the tube about one-eighth inch from the microphone end. Only 2 dB loss in sensitivity was measured with the resulting frequency response as shown in Fig. 50.

Smoothing the frequency response in this manner reduces "coloration" of the electronicly produced sound resulting in a more natural sound. The obvious disadvantage is the increased cost of production. These advantages and disadvantages should be considered in the final evaluation of the finished product.

Roanwell H 143/AIC Earphone: The H 143/AIC earphone unit is a rugged, moisture-proof, military type earphone unit designed for use in military communications systems. Although the unit is relatively bulky, its thin profile allows it to be conveniently fitted inside a hearing protector muff. The frequency response of the HI43/AIC earphone is flat from 20 Hz to about 6000 Hz when measured on an NBS 9-A coupler. These features of the H143/AIC earphone makes it suitable for consideration as a component of the electronic hearing protector system.

As a first attempt at a practical mounting, the unit was fastened on the inside of the foam lining of the earcup similar to the mounting of the D 426 earphone used in the first model of the electronic hearing protector built by the USBM. The resulting frequency response (Fig. 52) measured on the artificial ear shows that this method of mounting is unsuitable.

The H143/AIC earphone was attached to a 3/8" thick, closed cell foam neoprene wall arranged to hold the earphone as close to the pinna as possible without touching. The resulting response is shown in Figure 53. The many peaks and dips in the response are a consequence of a poor seal around the neoprene wall. Improving the seal resulted in the curve shown in Figure 54.

A new wall was built using the same neoprene material but more care was taken to insure a better seal around the edge. The earphone was cemented on the back of the wall so that it radiated through a ½" hole cut in the center of the neoprene wall. This minimized the amount of hard surface in the sound cavity. A cross-section drawing of this installation is shown in Fig. 55.

Concern over the loss of attenuation that results from the large volume taken up inside the earcup by the H143/AIC earphone and the closed cell foam wall led to a redesign of the mounting. The closed cell foam was replaced by ¼ inch open cell flexible urethane foam of about 2 lbs/cu. ft. density and the microphone and electronic package were incased in more of the same open cell foam. A cross-section drawing of this mounting is shown in Fig. 56. This design leaves as much as possible of the earcup volume open to maintain low frequency attenuation properties while at the same time holding the components securely in place. A muff seal was installed to allow more room for the pinna because of the space occupied by the foam.

The frequency response of the H143/AIC earphone mounted on the 3/8 inch closed cell foam wall is shown in Fig. 57. The response over the working range of 300 Hz to 3000 Hz is quite uniform and can be expected to result in excellent signal quality, however, the volume taken up by the components and the closed cell foam wall are known to affect the low frequency attenuation of the hearing protector and can be expected to degrade protector performance.

The response of the H143/AIC earphone in the open cell foam mounting is shown in Fig. 58. Although the response is not as uniform as in the closed cell mounting the rising response at 2800 Hz should not degrade intelligibility.

The attenuation of the complete system as a hearing protector, designed with open cell foam, was measured by the real ear method and by the artificial head method. A binaural mock-up of the hearing protector system was assembled in a pair of helmet-mounted muffs. Figure 59 shows the real ear attenuation of an unmodified muff-type protector. As expected, a certain amount of loss in attenuation is caused by the reduced earcup volume in the mock-up. However, the reduction in performance is probably due to the volume taken up by the components and not the presence of the open cell foam. The real ear data is listed in Table I for reference. (The artificial hear attenuation data in Fig. 59 was measured with a General Radio 1925 one-third octave multifilter and 1926 multichannel RMS Detector because the B & K equipment was inoperative at the time.)

1.3.8 Conclusions: The Knowles BL-1671 microphone has satisfactory frequency response and its small size is desirable for use in an electronic hearing protector. If desired, the response can be smoothed out considerably by suitable acoustic coupling. The Roanwell H143/AIC earphone has good frequency response but is rather bulky and tends to degrade the performance of the hearing protector at high noise levels or when the electronic system is off. A satisfactory arrangement for installing the equipment in a hearing protector earcup is to suspend the components in open cell foam. A double thickness muff seal allows extra clearance for the pinna to insure a comfortable fit.

1.4 Intelligibility Tests of Some Output Limiting Methods

1.4.1 Introduction to Experiment I: The purpose of this study was to investigate the influence on speech intelligibility of two output limiting systems proposed for use in the design of electronic hearing protectors. One of the systems evaluated provided output limiting by amplitude compression of the signal. The second system employed amplitude clipping as a means of signal output limiting. As a form of distortion, amplitude compression should demonstrate less reduction in speech intelligibility than amplitude clipping at high input levels. However, for the normal listener, speech remains highly intelligible under all but the most extreme forms of distortion.

To compare intelligibility of speech for different methods of output limiting, subjects listened to several lists of a multiple choice word test (MRT) recorded through two systems designed to limit the output of speech by amplitude compression and amplitude clipping.

1.4.2 Subjects: Ten (five female and five male) normal-hearing Pennsylvania State University graduate students served as subjects in this study. The subjects ranged in age from 21 to 24 years. None of the subjects reported otologic or upper respiratory infection at the time of the experiment.

Table I

Real-Ear Attenuation Measurements of Hearing Protector Assemblies

		125	250	500	1000	2000	3000	4000	6000	8000
		MSA - Comfo 500 (over-the-head c̄ headband) No. helmet								
18-26 July	$\bar{X}$	13.0	19.8	26.2	32.2	35.1	41.2	47.9	38.2	36.3
1975	SD	2.9	2.6	2.2	2.4	2.4	2.5	2.6	3.8	4.0
		MSA - Comfo 500 c̄ Roanwell H143/AIC Earphone								
8-14 Oct.		10.0	13.7	24.9	32.5	34.3	37.1	42.5	39.2	32.1
1975		2.3	2.6	3.0	2.2	2.4	3.3	4.7	4.3	3.6

1.4.3 Stimuli: Test stimuli were six lists (1A, 1E, 1F, 3C, 3D, and 3E) of the Modified Rhyme Test (MRT) (House, et al., 1965). The commercial tape recordings (Kreul, et al., 196) of these lists (S/N = +30 dB condition) were passed through one of two distortion networks. Network 1 (Fig. 60) was a laboratory-designed Two-Stage Diode Clipper which provided output limiting of speech signals by amplitude clipping. Distortion network 3 (Fig. 60) consisted of a dbx 119 Dynamic Range Enhancer which provided output limiting by signal level compression. The bypass network (Network 2 Fig. 60) provided linear reproduction of the speech stimulus and served as a distortionless control.

The test circuit for these distortion networks is shown in Fig. 61. The distortion networks were fed from the low impedance output of an operational amplifier and terminated by the high impedance non-inverting input of a second operational amplifier. Attenuators were provided at the input and output to adjust the signal through the distortion network to a level typical of operation at noise levels of 90 dB and 100 dB.

The commercially recorded lists of the MRT were reproduced by a Crown (700 series) tape player through one of the distortion networks set at the 90 or 100 dB response level or through the no-distortion bypass network and re-recorded on magnetic tape using a Crown (800 series) two channel tape recorder. In this manner, one of the six MRT lists was reproduced through one of five devised listening conditions. The first listening condition consisted of the no-distortion control. The dbx compressor was set to simulate the output limiting characteristics it would have at a noise level of 90 dB for the second listening condition and then at a simulated noise level of 100 dB for the third listening condition. For the fourth and fifth listening conditions the Two-Stage Diode Clipper network was similarly adjusted to the 90 and 100 dB output response levels. Two tapes of the five listening conditions were made. The order of the listening conditions and the MRT lists used for each condition were randomized in the recording of the two tapes. It should be noted that four of the six chosen MRT lists were used in both tapes. However, none of these four lists were used for the same listening condition in both tapes.

The listening conditions, attenuator settings for the 90 dB and 100 dB response levels, and the MRT list used for each condition are shown in Table II for the two tapes produced.

1.4.4 Apparatus and Procedure: The taped speech stimuli were reproduced on a Crown (800 series) two channel tape player through an Ampex (Model AA620) loudspeaker at 60 dB SPL. The output from a white noise generator was routed through a second Ampex (Model AA620) loudspeaker to produce a 0 dB signal-to-noise ratio. The sound field was calibrated before each test session using a General Radio (Type 1565-A) sound level meter set on the C scale positioned at the location of the subject's head. The loudspeakers were placed on a table

Table II

System Test Conditions

Listening Condition	Attenuator Setting dB 1	2	MRT List
Tape 1:			
Control (no distortion)	0.	11.	1A
dbx 90 (amplitude compression at 90 dB)	10.	0.	1F
Diode 90 (peak clipping at 90 dB)	10.	1.	1E
dbx 100 (amplitude compression at 100 dB)	0.	0.	3D
Diode 100 (peak clipping at 100 dB)	0.	10.	3C
Tape 2:			
Control	0.	11.	3E
Diode 90	10.	1.	3D
dbx 90	10.	0.	3C
Diode 100	0.	10.	1A
dbx 100	0.	0.	1F

against one wall of the test suite. The subjects were seated five feet in front of the mid-point of the loudspeaker array at approximately 0° azimuth.

Five subjects listened to Tape 1 and five subjects listened to Tape 2 of the five recorded test conditions.

1.4.5 Results: The mean MRT scores obtained by the subjects for each of the five listening conditions are presented separately for Tape 1 and Tape 2 in Table III. The pooled means, variances and standard deviations for both tapes are also shown in Table III. In all statistical tests, the probability of a type I error (α) was set at .05.

Results of the F_{max} test (Games and Klare, 1967) disclosed a significant difference (F_{max} = 30.953) among the variances for the five listening conditions pooled across the ten subjects. Specific F_{max} tests were performed on the five listening conditions to determine if the MRT scores obtained for any of the distorted listening conditions differed significantly from the no-distortion contol condition. The results of this analysis revealed that the MRT scores for the Diode 100 listening condition demonstrated significantly (F_{max} = 17.799) greater variance than the no-distortion control condition. Comparison of the variances for the Diode 100 and dbx 100 listening conditions indicated a significantly (F_{max} = 8.666) greater variability in the MRT scores for the Diode 100 condition than for the dbx 100 condition.

1.4.6 Conclusions I: The purpose of this study was to investigate the influence on speech intelligibility of two systems for output limiting proposed for use in the design of electronic hearing protectors. No significant difference in the means of the MRT scores for the five listening conditions could be demonstrated. However, there was considerable variability in the MRT scores obtained for the Diode 100 condition. The MRT scores for the Diode 100 condition ranged from 46% to 84% with a mean of 68.6% while the scores for the dbx 100 condition ranged from 62% to 88% with a mean of 75.4%. These results indicate that the mean intelligibility of the Diode 100 and the dbx 100 listening conditions is approximately the same. That is, in general, amplitude compression of speech is not superior in intelligibility to amplitude clipped speech at high input levels. However, the wide variability in subject scores for the Diode 100 condition points out the fact that, for some individuals, amplitude clipped speech introduces a sufficient amount of distortion to disrupt intelligibility. The variability effects of the Two-Stage Diode Clipper network may be expected to be more pronounced for older listeners with sensorineural hearing losses than for the normal-hearing young listeners used in this study. The high incidence of industrial workers with noise induced hearing losses, and the age range of these individuals, necessitates considerations of these factors in the design of electronic hearing protectors. Such protectors should provide the highest level of speech intelligibility for the greatest number of individuals who must use them.

Table III

Listener Test Scores

Listening Condition	Tape 1 mean	Tape 2 mean	Tape 1 mean	Tape 2 variance	S.D.
Control	66.8%	70.4%	68.6%	63.337	7.960
dbx 90	68.8%	75.2%	72.0%	36.444	6.036
dbx 100	73.2%	77.6%	75.4%	127.226	11.279
Diode 90	72.8%	72.4%	72.6%	45.377	6.736
Diode 100	76.4%	60.8%	68.6%	1128.04	33.586

1.4.7 Introduction to Experiment II: In Experiment I, it was found that the Diode 100 listening condition produced a significantly wide variability in MRT scores while the dbx 100 listening condition did not. However, in comparing the mean MRT scores for these two conditions, the dbx network was not superior to the Diode network in intelligibility. It was felt that the wide variability in MRT scores obtained for the Diode 100 listening condition could be reduced by filtering the speech passed through the Two-Stage Diode Clipper network. The harmonics produced by amplitude clipping may reduce the intelligibility of speech at high input levels for some but not for all listeners. Filtering of amplitude clipped speech should serve to decrease the number of harmonics in the output signal and reduce the wide variability in speech intelligibility. The purpose of this experiment was to investigate the effect of filtering on intelligibility when speech was passed through a Two-Stage Diode Clipper network. As in Experiment I, the amplitude clipped speech was reproduced through a high fidelity tape player. To simulate the bandwidth characteristics of a typical communication system, a 300-3000 Hz filter was placed in the circuit between the tape reproducer and the amplifier-speaker unit.

1.4.8 Stimuli: The test stimuli were the three taped MRT lists representing the Control, Diode 90 and Diode 100 listening conditions of Tape 2 from Experiment I.

1.4.9 Apparatus and Procedures: The same apparatus used in Experiment I was also employed in Experiment II. In addition a 300-3000 Hz passive speech filter (18 dB/octave roll-off) (Fig. 62) was connected between the output of the tape player and the input to the amplifier-speaker. The filtered speaker output was again calibrated to 60 dB SPL with a signal-to-noise ratio of 0 dB.

Four randomly chosen subjects who had participated in Experiment I served as listeners in Experiment II. The subjects listened to the three filtered MRT lists from Tape 2 representing the Control, Diode 90 and Diode 100 conditions.

1.4.10 Results: Individual subjects MRT scores for the filtered listening conditions are presented in Table IV. For comparison purposes, the unfiltered MRT scores obtained by each subject in Experiment I for the Control, Diode 90 and Diode 100 conditions is also shown in Table IV. Table V shows the mean unfiltered and filtered MRT scores for the subjects in Experiment II and the mean unfiltered MRT scores obtained by the ten subjects who served as listeners in Experiment I for the Control, Diode 90 and Diode 100 listening conditions.

1.4.11 Conclusions II: No statistical analysis was performed on the data obtained for Experiment II. In viewing the individual subject scores for the Control and Diode 90 listening

Table IV

Comparison of Test Scores for Filtered and Unfiltered Listeining Conditions

Listening Condition	Subject 1		2		3		4	
	unfiltered	filtered	unfiltered	filtered	unfiltered	filtered	unfiltered	filtered
Control	68	68	54	72	60	74	74	76
Diode 90	72	76	64	70	66	70	64	72
Diode 100	76	60	66	58	80	62	58	72

Table V

Mean Test Scores for Filtered and Unfiltered Listening Conditions

Listening Condition	unfiltered - 10 subjects (Exp. I)
Control	68.6
Diode 90	72.6
Diode 100	68.6
	unfiltered - 4 subjects (Exp. II)
Control	64.0
Diode 90	66.5
Diode 100	70.0
	filtered - 4 subjects (Exp. II)
Control	72.5
Diode 90	72.0
Diode 100	63.0

conditions, it can be seen that the subjects obtained the same score or demonstrated improved MRT scores for the filtered as compared to the unfiltered condition. The MRT scores remained the same for the unfiltered and filtered Control condition in one subject. Improved filtered versus unfiltered Control scores for the three remaining subjects ranged from 2% for subject 4 to 18% for subject 2. The mean improvement between the unfiltered and filtered Control condition for the four subjects of Experiment II was 6.5%. When the mean filtered Control score for the four subjects of Experiment II is compared to the mean unfiltered Control score for the ten subjects of Experiment I, there was an improvement of only 3.9%. All four subjects demonstrated improved MRT scores for the filtered as compared to the unfiltered Diode 90 listening condition. The amount of improvement ranged from 4% for subjects 1 and 3 to 8% for subject 4 with a mean difference of 5.5%. There was no difference between the filtered and unfiltered Diode 90 conditions when the filtered mean score is compared with the unfiltered mean score obtained by the ten subjects of Experiment I.

Although there were some intelligibility differences between the unfiltered and filtered Control and Diode 90 listening conditions for individual subjects, overall, no differences or only slight differences were demonstrated. It is difficult to account for individual differences, however, the general finding of no improvement between the unfiltered and filtered Control and Diode 90 listening conditions should be expected. As was shown in Experiment I, the Diode 90 condition did not differ significantly from the Control condition in either the variance or the mean. That is, the distortion of the Diode 90 listening condition did not produce a reduction in speech intelligibility over the no-distortion Control. Further, filtering should not change this result since most communication systems have an effective bandwidth corresponding to the 300 - 3000 Hz filter used in Experiment II.

Of major concern in Experiment II was the difference between the filtered and unfiltered MRT scores for the Diode 100 listening condition. Three subjects demonstrated a decrease and one subject showed an increase in MRT scores for the filtered condition as compared to the scores for the unfiltered condition. The mean reduction in the MRT scores of subjects 1, 2, and 3 was 14% while subject 4 demonstrated an improvement of 14%. A mean difference of 7% was found between the unfiltered and filtered Diode 100 condition for the four subjects of Experiment II. This difference is reduced to 5.6% when the mean unfiltered MRT score for the ten subjects of Experiment I is compared with the mean filtered MRT score for the Diode 100 condition. The results appear to indicate that intelligibility is not largely improved by filtering speech reproduced through an amplitude clipping network like the Two-Stage Diode Clipper used in this study. However, it was hypothesized that filtering would reduce the wide variability in scores obtained for the unfiltered Diode 100 listening condition.

Filtering produced a reduction in variance for all three listening conditions particularly for the Diode 100 condition. However, score variability continued to be greatest for the Diode 100 condition. There was also a decrease in the variance between the unfiltered scores for the ten subjects of Experiment I and the four subjects of Experiment II for the Diode 90 and Diode 100 conditions. The variance remained approximately the same for the two unfiltered Control conditions.

The results of Experiment II indicated that filtering did not improve the mean MRT score obtained for the Diode 100 listing condition; however, it did produce a reduction in score variability over the unfiltered Diode 100 condition.

2. TESTS OF ACOS A 9000/2 HEARING PROTECTOR

2.1 Introduction: The ACOS A 9000/2 hearing protector is an active, muff-type hearing protector commercially available in Great Britain from Cosmocord Limited, ACOS works. The muff uses fluid-filled cushions and is shaped to fit under a hard hat. It operates on the same principles as the USBM electronic hearing protector system except that the Lp under the muff is limited to 85 dB (according to the manufacturer) by the action of the electronics.

2.2 Tests of the Hearing Protector System: The ACOS hearing protector, with the electronics turned off, was placed on the artificial head in a diffuse sound field. The spectra of the sound under the muff for fields of 110 dB and 80 dB are shown in Figs. 63 and 64 respectively. The diffuse field spectra measured with the artificial head without the muff in place was adjusted to be flat from 63 Hz to 10,000 Hz.

The influence of the electronic system on the sound under the muff is shown in Figs. 65 through 70 for sound fields from 115 dB to 70 dB. The peak in the 2000 and 2500 Hz band is due in part to the position of the microphone with respect to the surface of the artificial head and probably is not as pronounced on a real ear. (Note that the peak is much lower than that obtained in Fig. 17.) These graphs have been combined in Fig. 71 to show the attenuation of the muff as the sound field is changed. It should be remembered that the attenuations shown here are those measured on the artificial head and are generally greater than those measured on a real ear.

The response of the electronic system alone was also measured with the B & K hearing aid test box. The plug-in electronics package with microphone attached was placed inside the box, and the earphone leads were brought out to the earphone and earcup mounted on the flat plate coupler on the artificial ear. Figs. 72 through 76 show the frequency response of the electronic system for sound pressure levels from 100 dB to 60 dB. Curves for right and left electronics (designed R and L after the figure number) are presented on the same pages for comparison. Frequency response limitations of the test box are such that the trace should be disregarded below 100 Hz. A summary of the responses for the right side system is presented in Fig. 77. Again, the peak at 2400 Hz is exaggerated by the microphone position in the flat plate coupler.

2.3 Response of Electronic System Components: With the ACOS earphone in the ACOS earcup mounted on the flat plate coupler, the unit was driven from the 6 ohm output from the B & K 1022 beat frequency oscillator. The resulting frequency response is shown in Fig. 78. This curve may be compared with the corresponding curve for the USBM-MSA earphone in Fig. 45. Note the general downward slope toward high frequencies.

The ACOS microphone was removed from the system and supported in the center of the diffuse field. The signal was amplified by a 20 dB gain laboratory pre-amplifier to produce the response curve in Fig. 79. Note in this curve the general upward slope toward high frequencies. This upward slope should tend to compensate for the downward slope of the earphone to produce a flat response; however, the overall response shown in Fig. 80 indicated that the earphone response is dominant. It is not apparent whether their responses were intended to be complementary or whether there is a reason for the upward sloping response of the microphone. It may be that the response of the microphone is unnecessarily compensated for in the amplifier circuit.

Input and output connections were added to the mother board of the ACOS A 9000 electronic hearing protector as shown in Fig. 81. It was necessary for the unit to be powered by the internal mercury cell to eliminate circuit oscillation. The acoustic output from the earphone was measured with a B & K 2204 SLM equipped with the B & K artificial ear and a flat plate adaptor for circumaural earphones.

The waveforms observed at pin 3 of the integrated circuit (SIGNAL OUTPUT point in Fig. 81) are shown in Fig. 82, A through E for various levels input signal. The upper half of the oscilloscope screen was calibrated to show the clipping action of the amplifier. The mercury cell used in this test measured 1.3 volts under load. This same calibration is maintained for the waveforms of Fig. 82, A through E. Figure 82A shows the waveform for 70 dB Lp acoustic output. The input signal level was then increased in 5 dB steps as indicated for Fig. 82, B through E.

The pure-tone transfer characteristics of the ACOS A 9000 hearing protector is plotted in Fig. 83 for various values of electrical input injected as indicated in Fig. 81. Because of the action of the limiting system it is safe to assume that an acoustic input would also produce an identical curve. (Note that the manufacturer claims that the ACOS A 9000 limits Lp at the ear to 85 dB.)

Figure 84 shows the frequency response measured as in Fig. 1 with pure-tone electrical input and acoustic output. The noise response of the ACOS electronic system was tested by injecting pink noise electrically as in Fig. 81 and measuring the acoustic output as the noise level was increased. Although the absolute input-output curve cannot be determined by this method, the transfer characteristics for an electrical input is shown in Fig. 85. This curve shows that the sound level under the muff reaches "maximum" then decreases rapidly as noise level input increases. The desirability of this effect was discussed in section 1 of this report. From this curve it appears that the electronically introduced sound pressure level under the muff reaches a maximum of 88 dB (A). Using an R-factor of 25 dB (calculated from real-ear attenuation data) it can be predicted that the ACOS A 9000 protector should limit the wearer to 90 dB (A) in sound fields up to 110 dB(A).

2.4 Hearing Protection of the ACOS Muff: With the electronics turned off, the ACOS A 9000 hearing protector was tested for real-ear attenuation according to the USASI Standard Z 24.22 1957 Method for the Measurement of the Real-Ear Attenuation of Ear Protectors at Threshold. The tabulated data and results appear in Table VII. The mean attenuation is compared with the manufacturer's advertised curve in Fig. 86. The manufacturer's curve (See ACOS Spec Sheet) shows remarkably good agreement with the real-ear measurements considering the fact that their curve was measured "with a flat faced hard wall artificial ear with simulated skin compliance." With the electronics turned on, the ACOS A 9000 hearing protector was placed on the EAL artificial test head in the diffuse field room. The acoustic signal measured at the artificial test head microphone is plotted in Fig. 6 as a function of sound field level in the test room. The R-factor of this protector was measured to be 24 dB by turning off the electronics with the sound field set at 90 dB (A). With this information the high level performance can be predicted by drawing a 45 degree slope line through the OFF point at 90 dB (A) input. The predicted high level performance should be asymptotic to this 45 degree line as indicated by the dotted extenstion of the measured curve.

2.5 Conclusions: Figure 82, A through E demonstrates that the limiting action of the ACOS A 9000 electronic hearing protector system is accomplished by amplifier saturation. The limiting level is, therefore, a function of the battery voltage and the sensitivity of the earphone unit. The unit appears to successfully limit the sound reaching the ears through the electronic system to 85 dB Lp.

Table VI

ACOS (A 9000) Active Muff: Anechoic Test (24-27 June 1975)

	125	250	500	1000	2000	3000	4000	6000	8000
1M	6	13	26	40	40	41	43	34	36
1M	6	13	26	38	38	43	47	34	40
1M	10	11	24	38	36	43	47	36	36
2F	10	13	25	28	44	42	50	36	40
2F	12	11	23	31	38	42	48	34	34
2F	10	11	19	32	40	44	48	32	36
3M	12	12	18	26	38	41	47	45	37
3M	8	12	16	30	34	43	47	41	33
3M	8	12	18	31	38	45	43	39	37
4F	16	14	18	25	38	39	41	42	35
4F	16	12	18	30	36	39	39	40	33
4F	16	12	20	30	36	37	41	38	29
5M	10	12	20	32	35	42	42	39	36
5M	8	10	18	33	37	42	40	37	34
5M	8	10	20	31	35	42	42	39	34
6M	12	11	19	36	39	48	37	42	38
6M	12	11	17	32	37	46	41	40	34
6M	10	9	17	31	39	46	43	38	38
7M	8	10	17	30	33	40	43	36	35
7M	6	10	15	28	35	38	41	34	33
7M	8	10	17	38	35	40	41	36	35
8M	17	15	17	31	38	43	39	42	38
8M	15	15	17	31	32	45	39	44	38
8M	13	13	19	31	36	41	43	38	36
9F	13	13	21	34	40	44	45	42	37
9F	15	15	23	33	40	46	43	42	39
9F	11	15	23	36	42	46	43	40	39
10M	12	16	21	34	37	45	47	40	36
10M	10	14	19	35	37	45	43	42	40
10M	10	18	21	36	37	41	45	40	38
$\bar{x}$	10.9	12.4	19.7	32.0	37.3	42.6	43.3	38.7	36.1
S.D.	3.17	2.11	2.99	3.52	2.56	2.67	3.22	3.29	2.52

From Fig. 87 it appears that the ACOS A 9000 active hearing protector will limit the sound level as the wearer's ear to 85 dB (A) in sound fields up to about 105 dB(A). With the electronics off, the ACOS A 9000 protector can be used in sound fields up to about 109 dB (A). The electronic system apparently causes a reduction in useful noise level range of about 4 dB, however this disadvantage is offset by the fact that normal communications are provided at low sound levels without the necessity of removing the protector.

3. EVALUATION OF ACOS A 9173 HEARING PROTECTOR

3.1 ACOS A 9173 Attenuation Performance:

The acoustic performance of the ACOS A 9173 Peak Limiting Ear Defender Headset was measured under a variety of operating conditions. This headset is an electronically controlled hearing protector essentially identical to the ACOS A 9000 except for the physical mounting arrangement. The protector was tested on the artificial test head in the EAL diffuse field test room with the sound field adjusted to a pink noise spectrum from 125 Hz to 10 kHz (see Fig. 88). The attenuation of the protector was then measured by observing the sound spectrum received by the microphone in the artificial test head with and without the protector in place. The attenuation is then the difference between these two spectra. The attenuation spectrum of the ACOS A 9173 headset with the electronics turned off is shown in Figure 89. Measuring the A-weighted sound received, with and without the protector in place gives an estimate of the R-factor which, for this protector is found to be 33 dB.

The level regulating action of the ACOS Headset electronics may be seen in Figure 90 through 93. Figure 91 may be compared with Figure 89 to illustrate the effect of the electronic system on the frequency response observed under the muff. Figures 90 through 93 show that a telephone quality acoustic transfer (see Section 1.2.3) is achieved by the electronic system for all of the noise levels tested.

Attenuation tests of three different sets of the ACOS A 9173 Peak Limiting Ear Defender Headset are listed in Table VII. In these tests, the sound field in the room was set to the level indicated in the first column. The sound level in dB(A) measured under each of the muffs is indicated in the table. At 107 dB(A) the electronics was turned off to obtain a measure of the maximum attenuation that can be achieved with the ear defender headset. The last column shows the mean attenuation observed at each of the sound field levels.

The acoustic transfer function data in Table VII is plotted in Figure 94. The line marked "Electronics off" is based on the measured R-factor of 33 dB and the assumption that the attenuation is linear at all levels. The attenuation value obtained with the electronics on shows that the sound level under the muff should remain below 90 dB(A) for external noise levels up to 110 dB(A). Actual tests were not made between 107 dB(A) and 123 dB(A) because these levels are beyond the capability of the present test facility.

Table VII Attenuation of ACOS A 9173 Protectors

	Set #1		Set #2		Set #3		Mean	
Sound Field dB (A)	dB (A) Under Muff	Attenuation dB	dB (A) Under Muff	Attenuation dB	dB (A) Under Muff	Attenuation dB	Acoustic Transfer dB	Attenuation dB
67	60	7	62	5	62	5	61.3	5.7
77	70	7	72	5	72	5	71.3	5.7
87	79	8	81	6	81	6	80.3	6.7
97	85	12	87	10	87	10	86.3	10.7
107	88	19	90	17	89	18	89	18
Electronics Off								
107	74	33	74	33	73	34	73.7	33.3

3.2 Speech Discrimination Performance of ACOS A 9173 Protectors:

This study was conducted to compare speech discrimination ability of persons wearing various types of hearing protectors. The protectors of special interest to the U.S. Bureau of Mines are:

1. A conventional helmet-mounted, muff-type hearing protector.

2. A peak-limiting, electronic hearing protector.

Discrimination scores obtained without protectors were also measured as a control measure.

3.2.1 Subjects: A group of ten trained listeners, all having normal hearing sensitivity and ranging in age from 22-33 years participated in this study.

3.2.2 Equipment: A pink noise field was produced using a General Radio Random Noise Generator, Type 1390-B, equipped with a 1390-P2 pink noise filter. The output was shaped to compensate for room acoustics by using a B & K Graphic Spectrum Equalizer, model 124. The required sound levels were produced by amplifying the pink noise signal through a pre-amplifier, and a Crown DC-300 amplifier connected to three corner-mounted speakers in the test room. (See Fig. 95)

The spectrum was calibrated daily using a B & K Audio Frequency Spectrometer, Type 2113 1/3 octave band analyzer. A one-inch B & K 4144 microphone was placed in the center of the test room at a point duplicating the position of the subject's head during testing. The sound field was adjusted, using the spectrum equalizer, so that each 1/3 octave band from 125 -10kHz was within 1 dB of the sound pressure level at the 1kHz 1/3 octave band. The B & K 2331 spectrometer was then set on linear, and using the amplifier controls, the sound pressure level was adjusted to 105 dB with the attenuator dial set to 0 dB attenuation.

Speech discrimination was measured with recorded word lists reproduced on a Crown International 700 series tape deck connected through a shout filter (Michael, P. L. et al. 1973) and a power amplifier to a separate speaker in the test room. (See Fig. 95)

The voice channel was calibrated by placing the speaker one meter from the reference microphone in the center of the test room. Using the B & K Audio Frequency Spectrometer, the calibration noise on the tape was set to 85 dB linear by adjusting the power amplifier (with the shout filter out of the circuit). The attenuator dial was then set to produce the required signal-to-noise ratios.

3.2.3 Test Material: Lists (A, F, B, & E) of a taped presentation of the Modified Rhyme Test (MRT) were used as the test stimuli. The MRT is a six-alternative forced choice test that assesses word discrimination ability on both initial and final consonants. Each list consists of 50 test items presented in a sentence format.

3.2.4 Test Muffs: Two helmet mounted M-S-A Comfo 600 muffs and two ACOS Cosmocord electronic muffs were tested in the study.

3.2.5 Procedures: Subjects were each seated in the diffuse sound field facing the speaker through which the MRT test was presented. Each subject was tested under three randomized protector conditions: (1) while wearing a conventional MSA Comfo 600 muff; (2) while wearing an ACOS Cosmocord muff; and (3) while wearing no muff. Within each test condition subjects were exposed to four levels of speech and noise. (See Table VIII)

Thus, each subject completed a total of 12 separate MRT test forms which represented their discrimination ability in varying levels of speech and noise that are considered to be typically encountered in a mining environment.

In each test condition, all subjects were required to wear an M-S-A Comfo-Cap equipped with a helmet light and battery pack.

A ten minute rest period was given after completing the testing under each condition. A 20 minute rest period was required upon completion of the unprotected exposure to minimize the effects of temporary threshold shift on further testing.

3.2.6 Results: Results reported in the Analysis of Variance Summary shown in Table IX demonstrate that there is a non-significant interaction between the protector condition and the listening condition. That is, there are no significant differences found between speech discrimination scores measured with the various protector and listening conditions.

Differences between the hearing protectors were also non-significant indicating that speech discrimination scores were comparable for the same listening or signal to noise (s/n) conditions. In terms of speech perception, it made no difference whether the listener wore ear protectors or was unprotected.

A highly significant F-ratio was found for the listening condition factor. This indicates that for all protector conditions, as the difficulty of the listening condition was increased, speech discrimination scores decreased.

Similar results are found in a follow-up test by the T-method (P. 383, Glass and Stanley) which are reported in Table X. With this test, mean values of speech discrimination scores are averaged across the protector condition. It was found that the means of listening condition one (speech 65 dB; noise 35 dB) and two (speech 65 dB; noise 65 dB) are significantly different from one another and from the mean of condition three (speech 80 dB; noise 90 dB) and four (speech 95 dB; noise 105 dB). However the means of condition three and four are not significantly different from one another. These results indicate that the one factor that greatly influences the difficulty of the listening condition is the signal to noise ratio. Changing the s/n produced a significant difference in discrimination score means. Maintaining a constant s/n, while increasing only the intensity of both speech and noise produced very similar means.

Table VIII Signal levels used for measurement of speech discrimination with hearing protectors

Speech	Noise	s/n
1) 65 dB	35 dB	+30
2) 65 dB	65 dB	0
3) 80 dB	90 dB	-10
4) 95 dB (with shout filter)	105 dB	-10

Table IX Analysis of Variance

	Mean Squares	R-Ratio	Probability of Error
Protector Condition	111.4333	2.939	0.121
Error	37.91481		
Listening Condition	20824.22	110.636	0.000
Error	188.2222		
Interaction of Protector Condition and Listening Condition	35.65556	1.364	0.273
Error	26.13704		

Table X Mean Discrimination Scores Averaged Across Protector Condition

(1)	(2)	(3)	(4)
Speech 65 dB Noise 35 dB s/n +30	Speech 65 dB Noise 65 dB s/n 0	Speech 80 Noise 90 s/n -10	Speech 95 dB Noise 110 dB s/n -10
87.3%	58.8%	32.5%	31.6%*

* ____________________ indicates a non-significant difference between means.

Table XI reports discrimination means for the protector condition averaged across the listening conditions. It is evident that word discrimination scores were not dependent on whether an electronic muff, a conventional muff, or no muff was worn.

A summary table (Table XII) is provided showing the means for individual protector conditions and listening conditions.

It can be concluded from these data that the use of neither a conventional nor an electronic ear protector enhances or degrades speech perception. A person wearing ear muffs maintains the same ability to discriminate speech as a person wearing no muff, regardless of the levels of speech and noise to which they are exposed so long as the s/n ratio is constant.

Some final comments should be made in regard to our experience with the Cosmocord ear protector. Many of the participants in the study noted that the headband was uncomfortably tight, and many had difficulty in fitting the helmet over the earmuff because of the greater bulk caused by the head strap attachment. Subjects reported that the electronic system did not produce a "natural" sound but that the sound had a "crisp" quality similar to that of an intercom. The off-on toggle switch is an improvement over the push-on-push-off switch provided on the ACOS A 9000, but it is still vulnerable to accidental turn-off by bumping the protector.

3.3 Conclusions: The results of speech discrimination tests indicate that the ACOS A 9173 protector offers no advantage for voice communications over conventional hearing protectors or no protectors in a normal coal mine situation. The ACOS A 9173 protector, through its electronic system, may allow faint roof talk or speech from distant talkers to be heard that might not be heard while wearing a conventional hearing protector, although the electronic noise of the system will mask low frequency (<500 Hz) sounds lower in level than about 26 dB L_p.

The results of this study reinforce the recommendation made by EAL that hearing protectors be worn in a potentially dangerous noisy environment to prevent permanent hearing losses.

Table XI Mean Discrimination Scores for Protector Condition Averaged Across Listening Condition

Electronic Muff	Conventional Muff	No Muff
43.15%	50.40%	35.50%

Table XII Mean Discrimination Scores for Individual Protector and Listening Condition Combinations

	s/n = +30 Speech 65 dB	s/n = 0 Speech 65 dB	s/n = -10 Speech 80	s/n = -10 Speech 95
Electronic Muff	86.2%	58.8%	33.6%	34.0%
Conventional Muff	84.4%	56.6%	30.6%	30.2%
No Muff	91.4%	58.8%	33.2%	30.6%

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

4.1 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.

4.2 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 dis-crminiating 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.

4.3 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).

5. MODIFICATIONS TO THE EAL-USBM TRAINING COURSE

5.1 Training Course in Hearing Conservation: Revisions were made to the training course in Hearing Conservation for Coal Miners as requested by the U. S. Bureau of Mines. Minor revisions were made to the script and new drawings were made for visuals 1, 11, 24, and 33. A more suitable musical selection was chosen for use in setting reproduce level of the audio demonstration tape.

The hearing protector simulation network was rebuilt to conform to the mean attenuation curve of the five types of circumaural hearing protectors tested under a PHS study performed by EAL (Michael & Bolka, 1972). Demonstration material for this and two other segments was re-recorded using live voice instead of the original material recorded in the coal mines. This produced a better demonstration by improving the naturalness and the signal to noise ratio.

Sets of the revised materials have been distributed to former USBM recipients of the original training course produced as part of the 1974 research grant.

5.2 Automation of Hearing Conservation Training Course: At the request of the U.S. Bureau of Mines the Course in Hearing Conservation for Coal Miners (originally produced by EAL under USBM Grant G 0133026) was produced as a fully automated slide-lecture program. The script was recorded on magnetic tape by a professional reader. The special audio demonstration material supplied with the original script was assembled into the master tape at the appropriate points. Using a tape recorder with selective synchronization capability the slide control pulses were recorded on a separate tape track. The resulting master tape contains the audio program on Channel 1 (left channel) and the slide control cues on Channel 2 (right channel). The track configuration on the master tape is standard for two-track stereo-recording as illustrated in Figure 96. The master tape can be duplicated or copied into other track configurations as needed.

5.2.1 Slide Synchronization Systems: Slides are synchronized with the recorded narration by recording a tone pulse at the appropriate point in the narration where a slide change is desired. When the tape recording is played back, the tone pulse activates the slide change mechanism in the projector through a cue-tone detector circuit. Professional slide synchronizing equipment using this principle is manufactured by at least three major companies:

Spindler & Sauppé: This company manufactures a complete line of multi-media audio-visual control systems used for many industrial and entertainment productions. This simple system uses a 1000 Hz tone to control a single slide tray projector or a twin-projector dissolve system*. The narration is recorded on tape track one and the slide synchronizing

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*A dissolve system fades the lamp off on one projector and on in the second projector while slides are changed in the "off" projector. The result is a fade from one slide to another with no distracting image movement on the screen.

signals are recorded on track two. This arrangement minimizes interference between the control signals and the narration. The full frequency response of tape track one is then available for the audio program.

3M-Wollensak & Caramate: The 3M and Caramate slide synchronization systems use the same 1000 Hz tone to activate the slide change mechanism as the Spindler and Sauppé system but is primarily designed around a cassette recording system. While the control signals are available externally, the system is available only with the cassette recorder-player and consequently costs considerably more than the Spindler-Sauppé control unit alone. The control signals of the Spindler-Sauppe system are compatible with the 3M and Caramate slide synchronization systems and vice-versa. Cassettes produce for the 3M-Wollensak system and also compatible with the Caramate system.

DuKane: In the DuKane System slide changing is activated by a 50 Hz tone recorded on the narration track, consequently only a single-channel recording and reproduction system is necessary. However, the audio program must be filtered to suppress the 50 Hz control signal and, although this presents no difficulty when the audio program consists of speech only, the quality of any special audio demonstration material will be degraded by the limited frequency response. The 3M-control system can be switched to accommodate DuKane control signals.

5.2.2 Choice of Synchronization System for USBM:
The Spindler & Sauppé Uni-Q was chosen for synchronizing the slides and narration for the Course in Hearing Conservation for the following reasons:

1. The Spindler & Sauppé system is low priced, simple and reliable.

2. Master tapes can be produced on 1/4 inch reel-to-reel tapes for ease of editing and best quality.

3. Tape copies can be produced to be compatible with the 3M systems.

4. Control signals do not interfere with or limit the fidelity of the audio program.

5.2.3 Magnetic Tape Format: A two-channel (stereo) tape player system is necessary for presentation of this synchronized slide program. In the 1/4 inch reel version, the narration and special audio demonstration material is recorded on track one (left channel) of the tape and the slide control signals are recorded on track two (right channel). The master tape for the Course in Hearing Conservation is the standard two-track (stereo) format illustrated in Figure 96. Direct copies of this tape can be reproduced on any reel-to-reel stereo tape player using the standard two-track configuration.

If a four-track (sometimes called 1/4-track) stereo tape player is to be used, the master tape must be copied into this track format shown in Figure 97. A two-track stereo recording can not be reproduced satisfactorily on a four-track tape player (unless provision is made for shifting the position of the reproduce head). A properly recorded and reproduced four-track recording will give satisfactory results with this synchronized slide presentation.

Special copying equipment is necessary to produce cassette tapes compatible with the 3M and Caramate control systems. In these systems the track arrangement on the tape (Figure 98) is similar to the two-track format described in Figure 96. Track 1 corresponds to the normal monaural configuration for cassette tapes. Track 2 corresponds to the monaural track on side two of the cassette. This track configuration can be produced by copying the master tape into a monaural cassette using a professional copier such as a Pentagon High-Speed Cassette Copier Model C-1320. This machine copies both tracks of the two-track master tape into the required configuration on the cassette tape in one pass through the machine. Such a copier is available at the Pennsylvania State University.

While the standard stereo track configuration for cassette systems (Figure 99) qualifies as a two channel system, the quality of monaural reproduction from a single channel of a stereo cassette system will degrade the quality of this presentation. The use of a stereo cassette system is not recommended for this synchronized slide program.

5.2.4 Method of Presentation: For the best quality presentation of this automated slide program, the following equipment is recommended:

1. Rotary Tray Slide Projector such as Kodak Carrousel with capacity for 51 slides or more.

2. Two-track stereo tape player for 7" reel-to-reel 1/4 inch tapes.

3. High quality sound reproduction system.

4. Slide cueing system responsive to 1000 Hz cue tones such as Spindler and Sauppé Uni-Q.

This equipment should be interconnected as shown in Figure 100. Sequence and description of slides is shown in Table XIII. With the projector on, the title slide "Hearing Conservation for Miners" should be focused and adjusted to fill the screen. All slides are in horizontal rectangular format and have the same size aperture. Starting the tape player will then operate the entire program automatically. The first slide cue tone will switch in the second slide; no cue tone is provided for the title slide since it should already be on the screen at the start of the program. The program requires approximately 30 minutes and ends with the "End" slide left on the screen.

Table XIII Slide sequence for Automated Training Program

Slide Tray Position	Slide Number	Description
1	Title:	Hearing Conservation for Miners
2	B & T P-8	Miner on locomotive
3	B & T D-13	Three miners with roof-bolter
4	NCA 21A	Three miners, close-up, talking
5	EAL 5A	Noise-induced hearing loss (sign)
6	B & T	Miners emerging from elevator
7	NCA 16A	Miners in lunch area (crowd)
8	EAL 8A	Miner outline with ?
9	EAL 1	Ear drawing
10	EAL 2	External ear
11	EAL 3	Ear wax build-up
12	EAL 5	External ear infection
13	EAL 7	Rupture of eardrum
14	EAL 4	Spark damage to eardrum
15	EAL 6	Acoustic trauma
16	EAL 8	Middle ear
17	EAL 9	Photograph of middle ear
18	EAL 10	Middle ear infection
19	EAL 11	Blockage of eustachian tube
20	EAL 12	Otosclerosis
21	EAL 13	Inner ear
22	EAL 14	Vertigo
23	EAL 15	Photograph of first turn of cochlea - normal nerves
24	EAL 16	Hair cell group photograph
25	EAL 19	Noise induced hearing loss "Roar, Pop, Bang"
26	EAL 18	Photograph of first turn of cochlea - damaged nerves
27	NCA 343	Roof drilling operation - single operator
28	B & T 6-17	Continuous miner
29	EAL 21	Tinnitus
30	EAL 22	Recovery of hearing after exposure
31	EAL 23	Exposures away from work
32	EAL 24	Comparison of typical noise levels
33	EAL 25	Rifle shooter
34	EAL 26	Inner ear noise-induced hearing loss (Trombone)
35	EAL 27	Teen-ager with transistor radio
36	EAL 28	Reduction of hearing level from daily exposure
37	EAL 29	Audiograms
38	EAL 30	Presbycusis
39	EAL 29	Audiograms (duplicate)
40	B & T D-4	Shuttle Car
41	EAL 32	Miner with helmet-mounted protectors (drawing)
42	NCA 15A	Two miners--one with hearing protectors on helmet
43	EAL 33	Protection (dB) from cotton and properly fitted protectors
44	EAL 34	Removing muff seals
45	EAL 35	Washing muff seals
46	EAL 36	Replacing muff seals
47	EAL 37	Excess stretching of head band
48	EAL 38	Sealing over safety glasses

Table XIII (continued)

Slide Tray Position	Slide Number	Description
49	EAL 39	Noise-induced hearing loss (chart)
50	EAL 40	Miner kicking
51	End	

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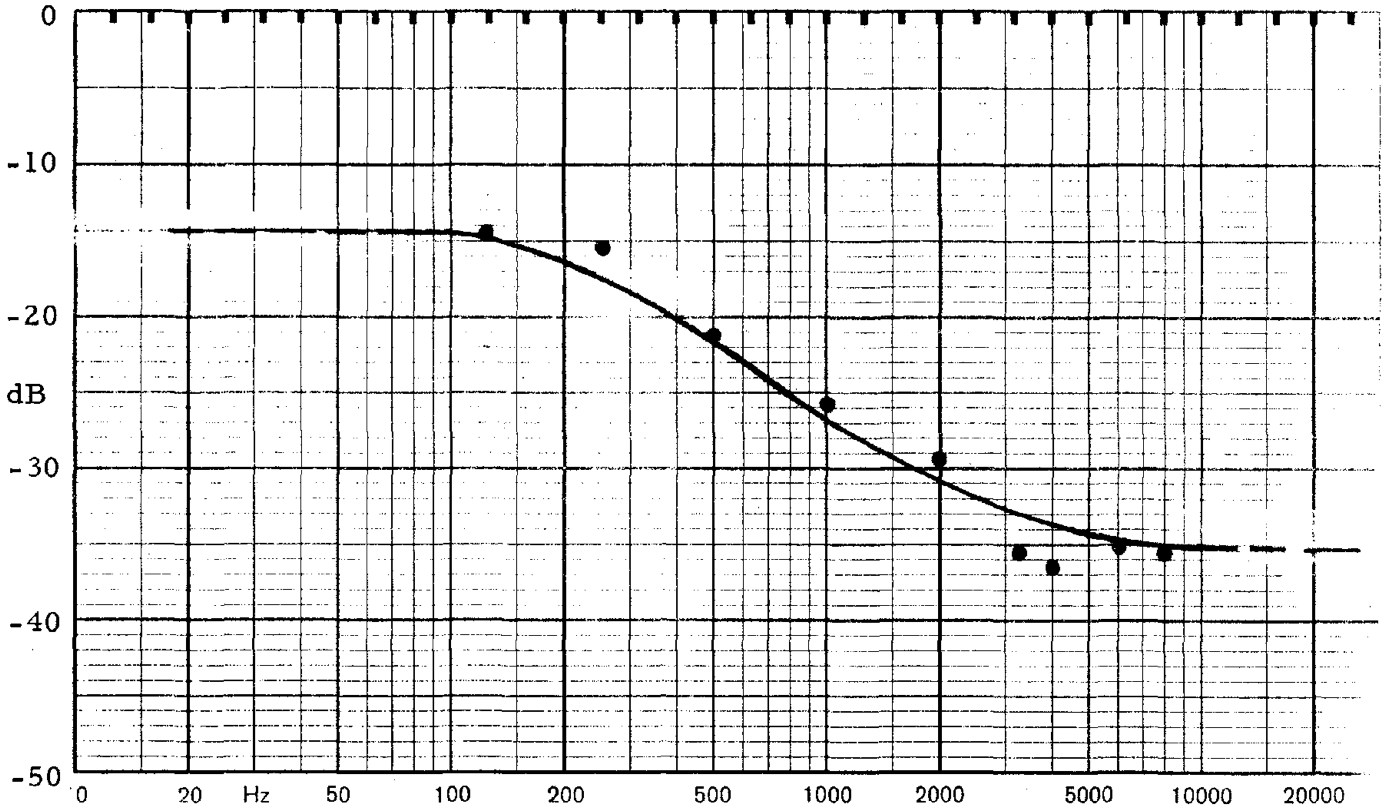


Fig. 1 Mean attenuation of five different types of hearing protectors.

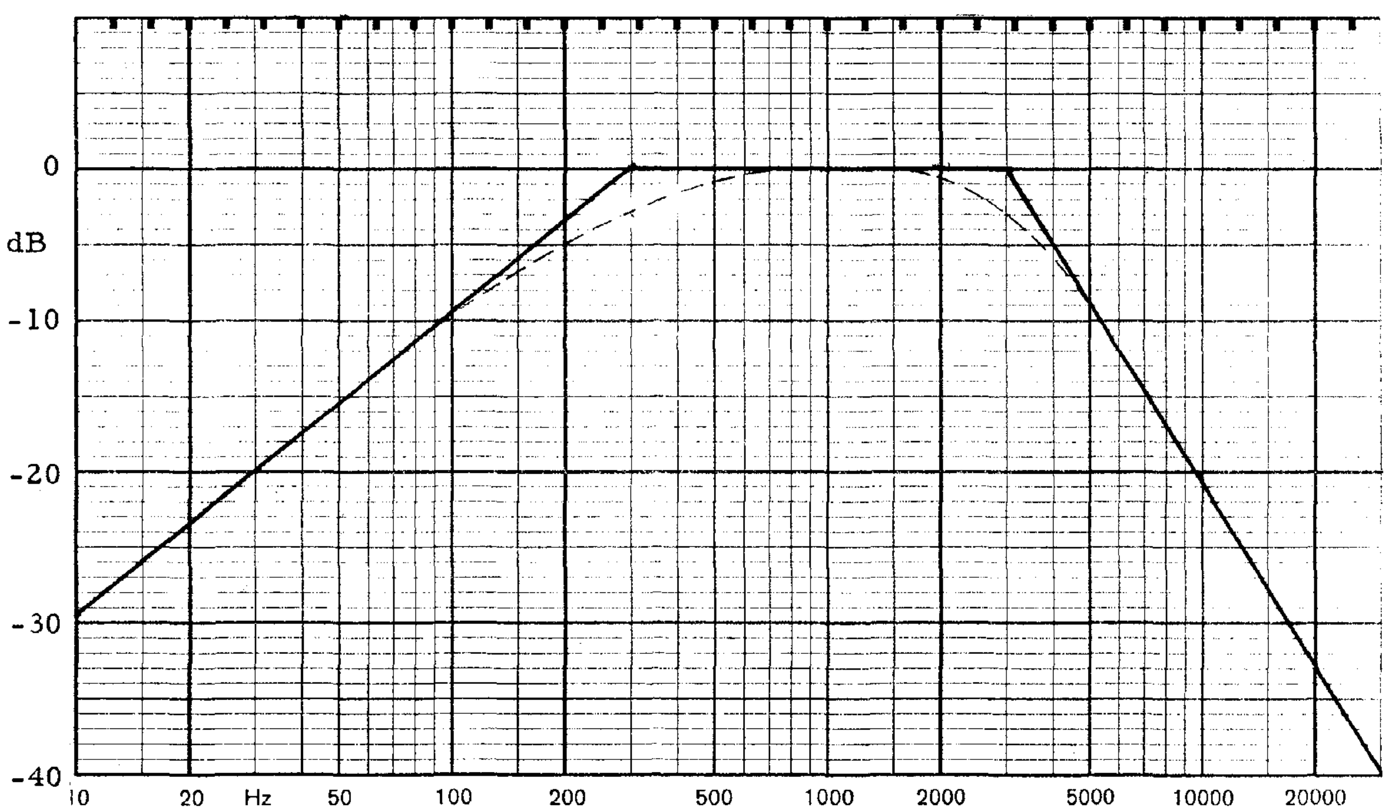


Fig. 2 Recommended frequency response for electronic hearing protector communication system.

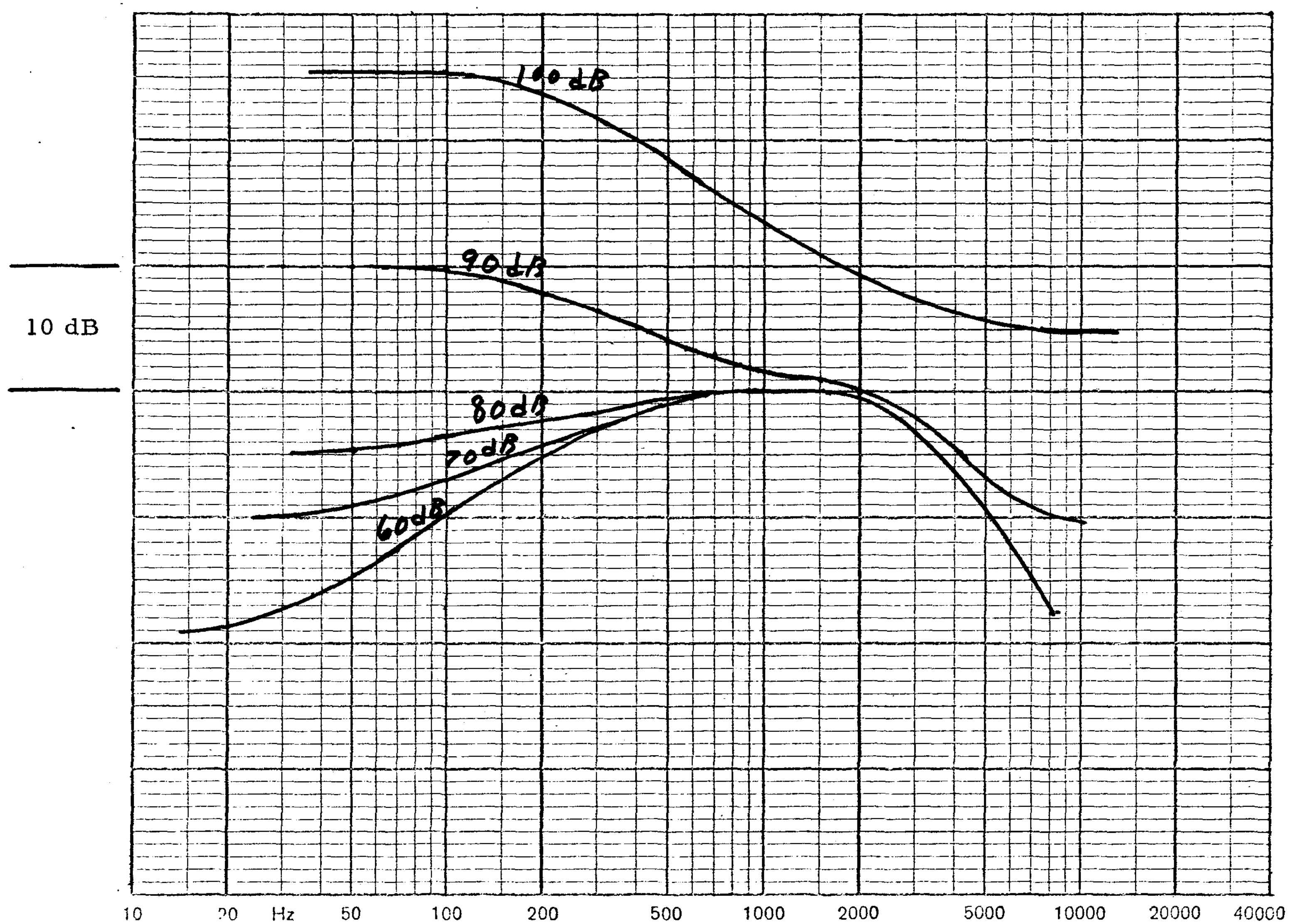
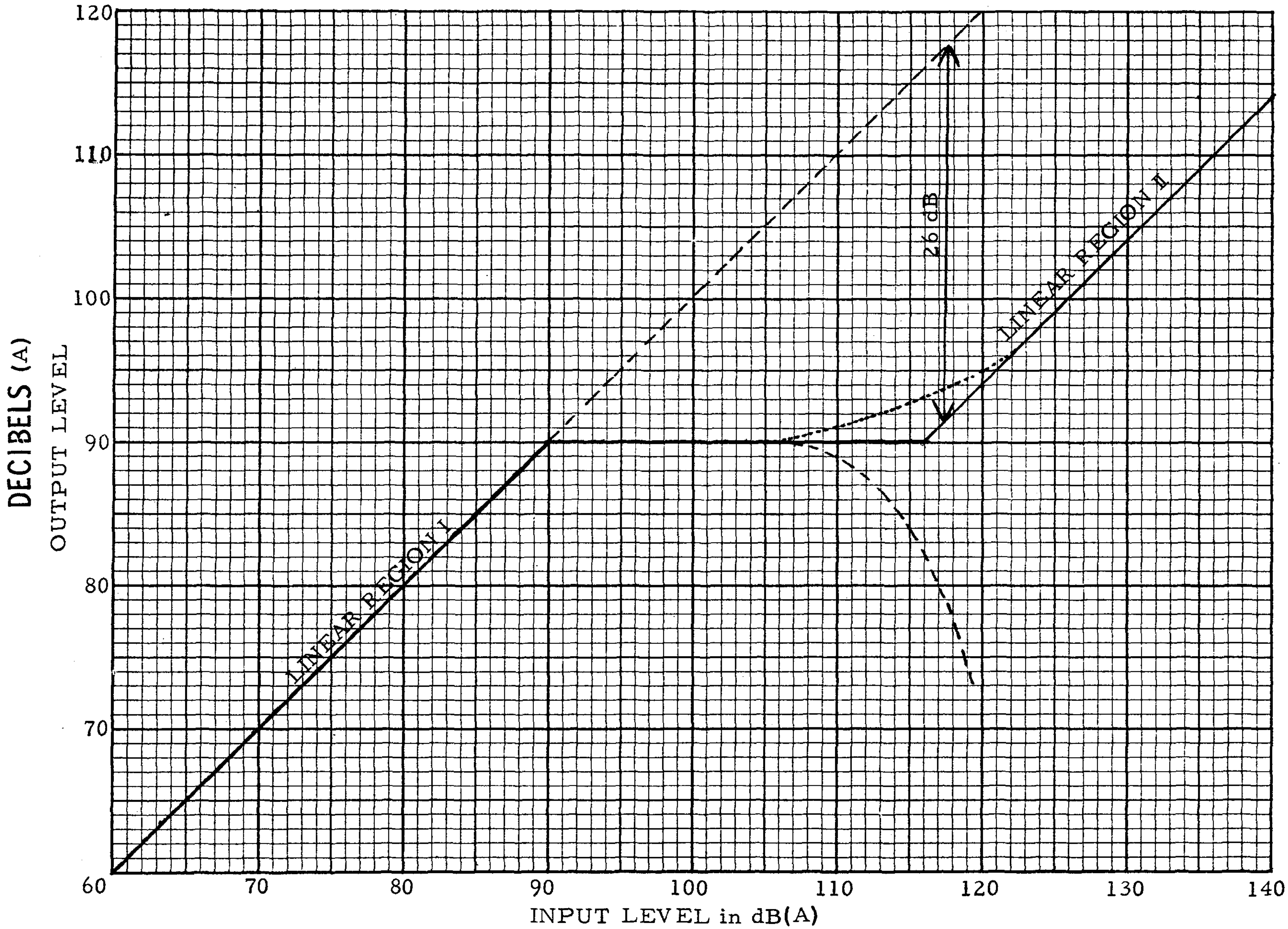


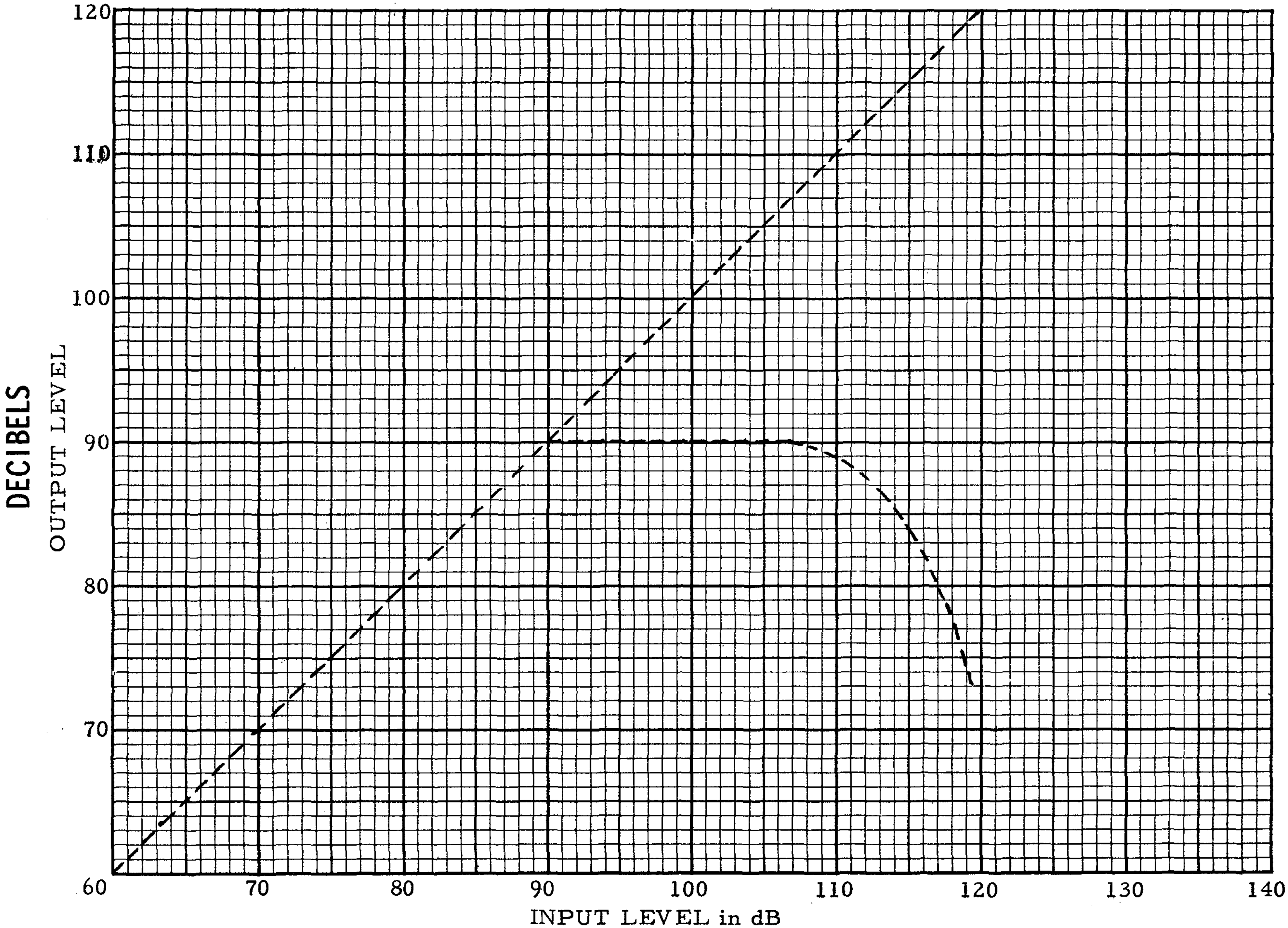
Fig. 3 Electronic hearing protector response for some representative ambient noise levels.

FIGURE 4



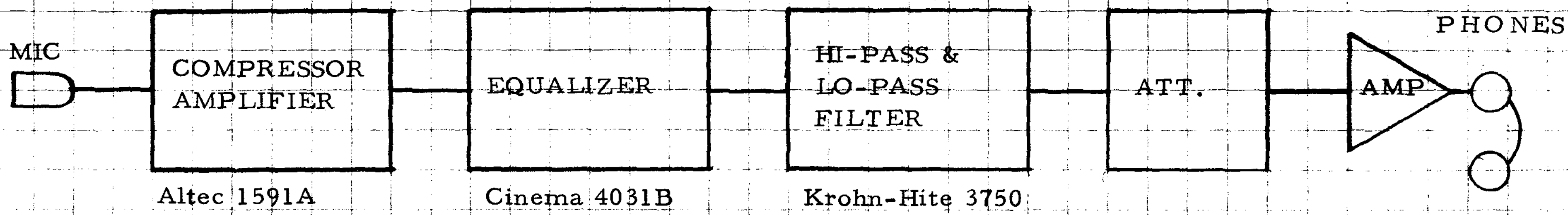
TRANSFER FUNCTION OF AN IDEAL ELECTRONIC HEARING PROTECTOR

FIGURE 5



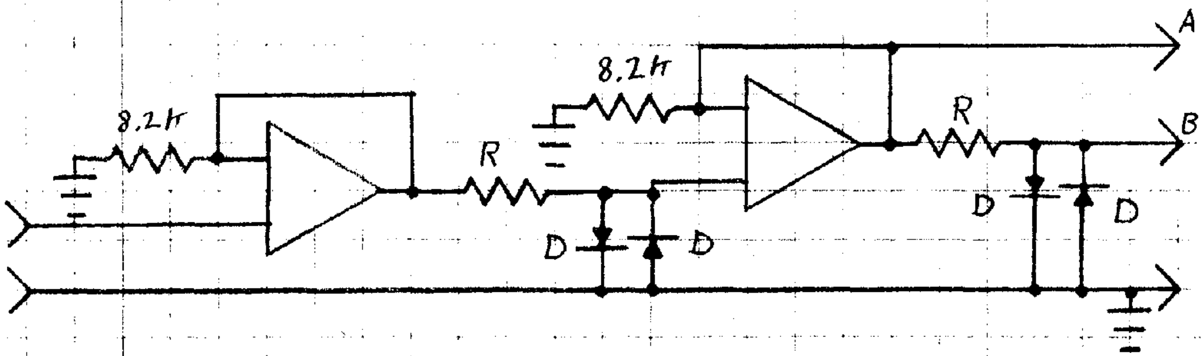
ELECTRONIC TRANSFER FUNCTION REFERENCE GRID

FIGURE 6

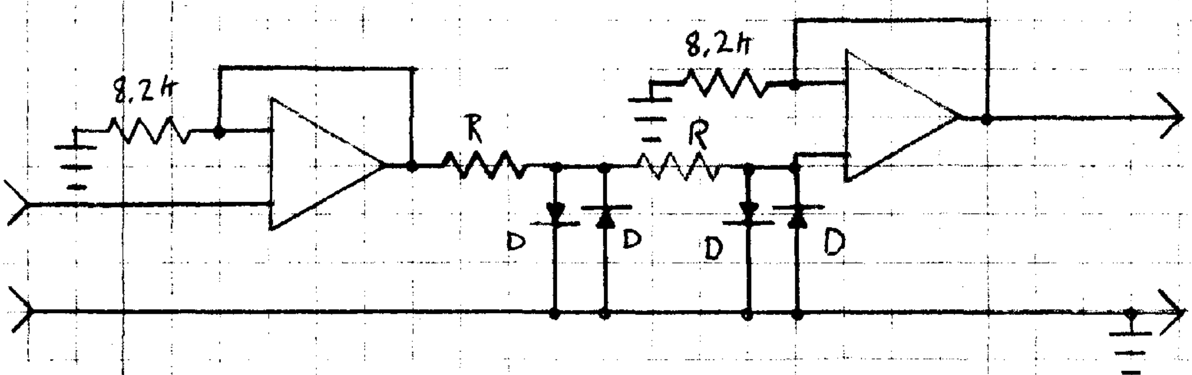


Block diagram of electronic ear protector transfer function simulator.

FIGURE 7

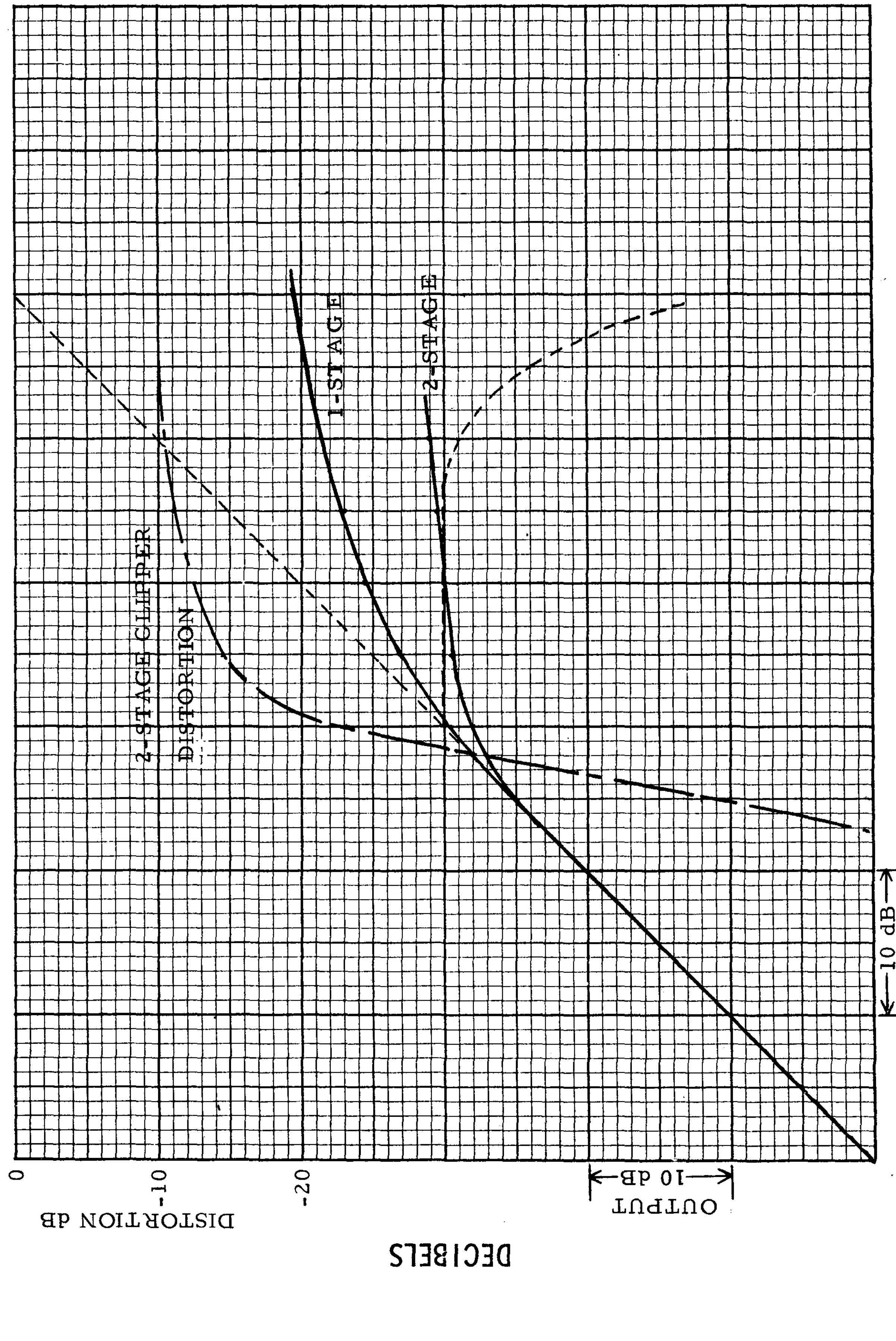


A. Single-Stage Diode Clipper
B. Isolated Two-Stage Diode Clipper



C.
Non-Isolated Two-Stage Diode Clipper

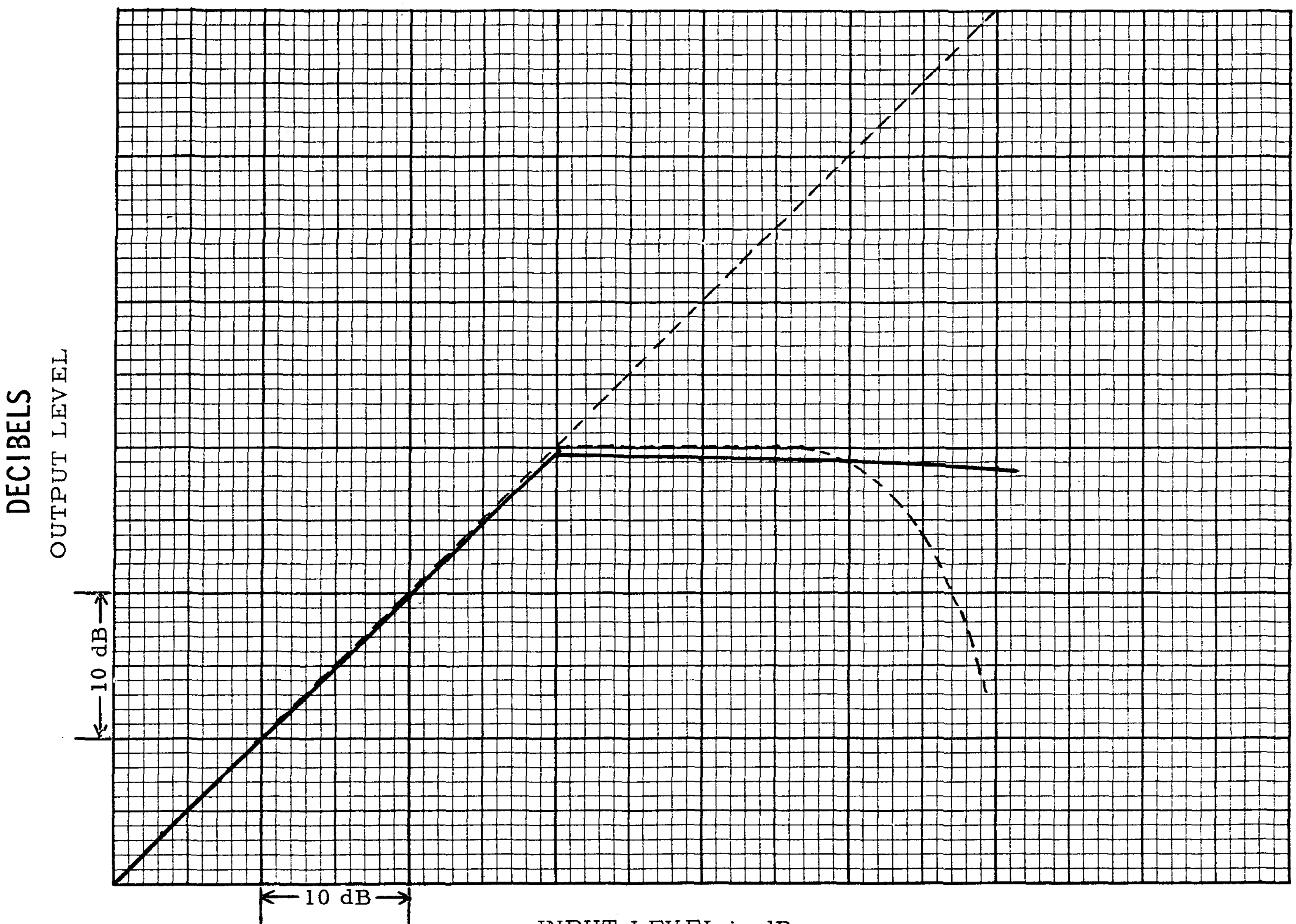
FIGURE 8



INPUT LEVEL in dB

TRANSFER CHARACTERISTICS OF DIODE CLIPPERS

FIGURE 9



TRANSFER CHARACTERISTIC OF dbx COMPRESSOR

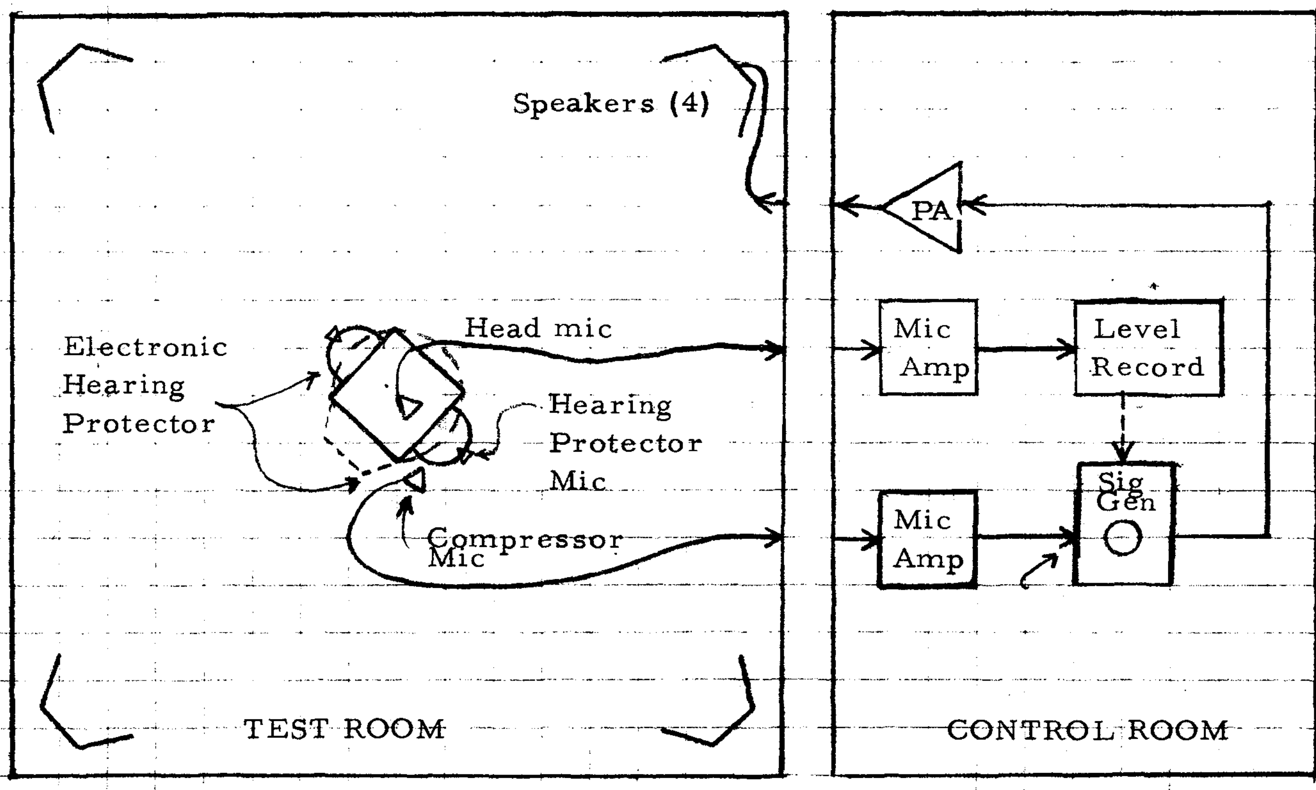


Fig. 10 Equipment arrangement for measuring pure-tone response of hearing protectors in diffuse field room.

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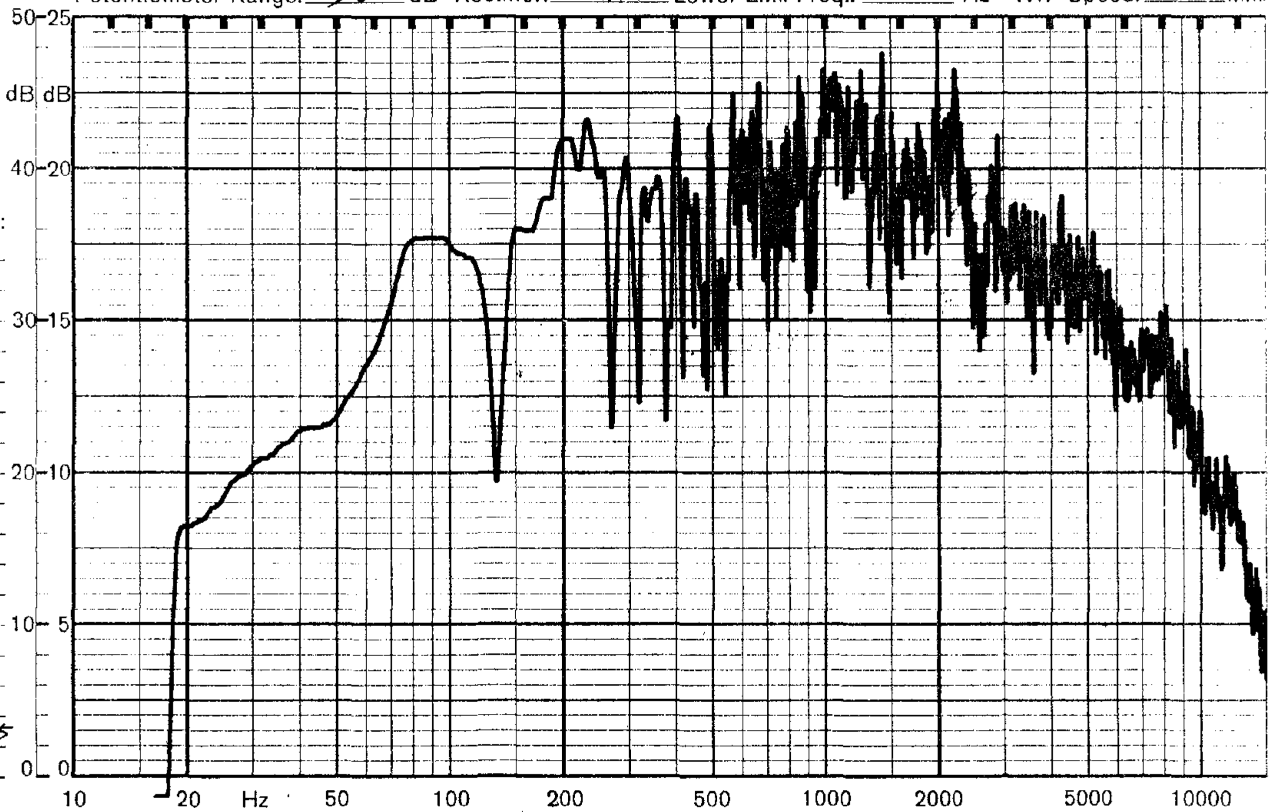
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 31,5 mm.

Fig. 11

Measuring Obj.: Sound Field 100 dB

Rec. No.: 1
Date: 27 May 75
Sign.: JJP



QP 1124 Multiply Frequency Scale by Zero Level: 60 dB (161

Brüel & Kjær Brüel & Kjær Brüel &

Brüel & Kjær Copenhagen

Potentiometer Range: 50 dB dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: mm/

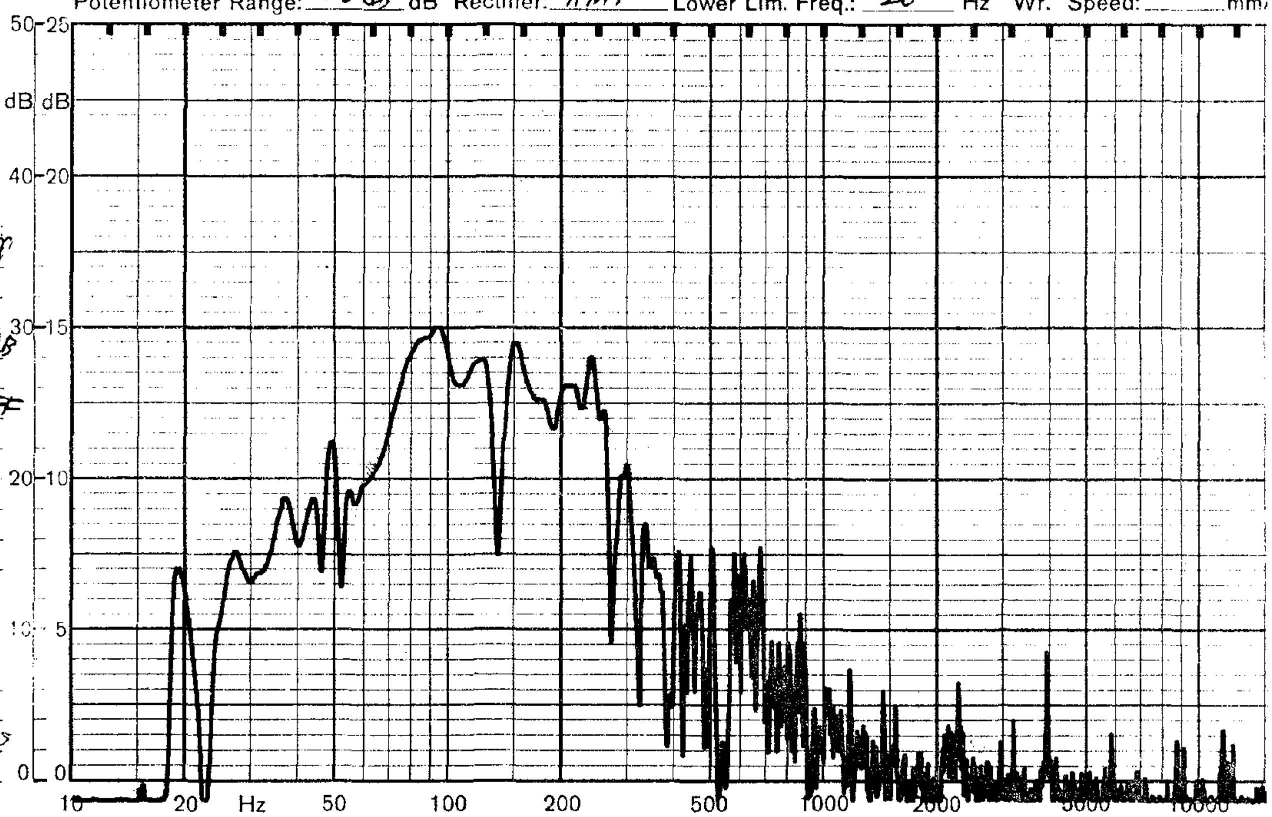
Fig. 12

Measuring Obj.: Electronic Muff USBM No1 Sound Field 100dB

Electronics OFF

Writing Speed 31,5 mm/Sec
Paper Speed ,3 mm/sec

Rec. No.: 2
Date: 27 May 75
Sign.: JJP



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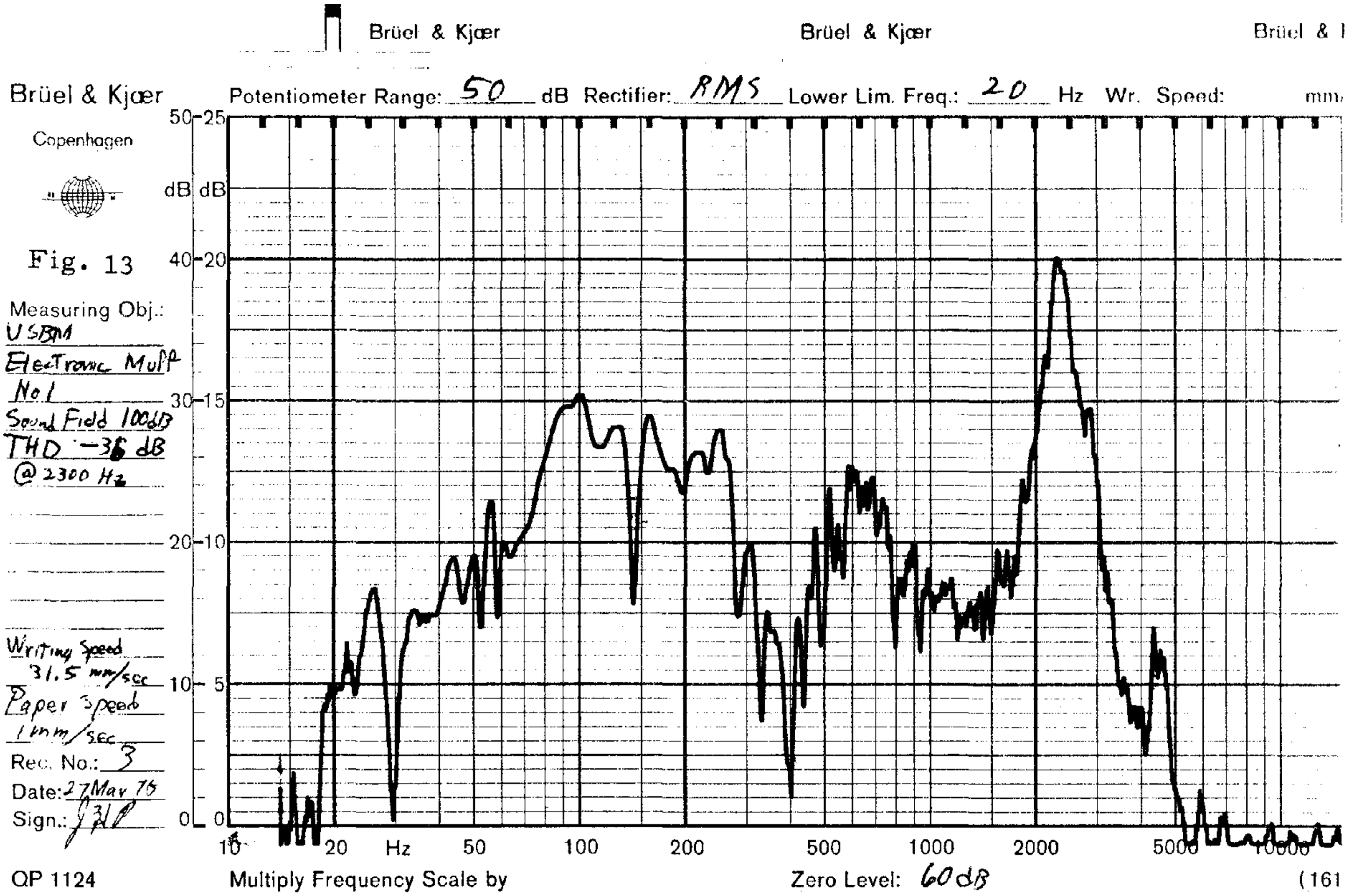
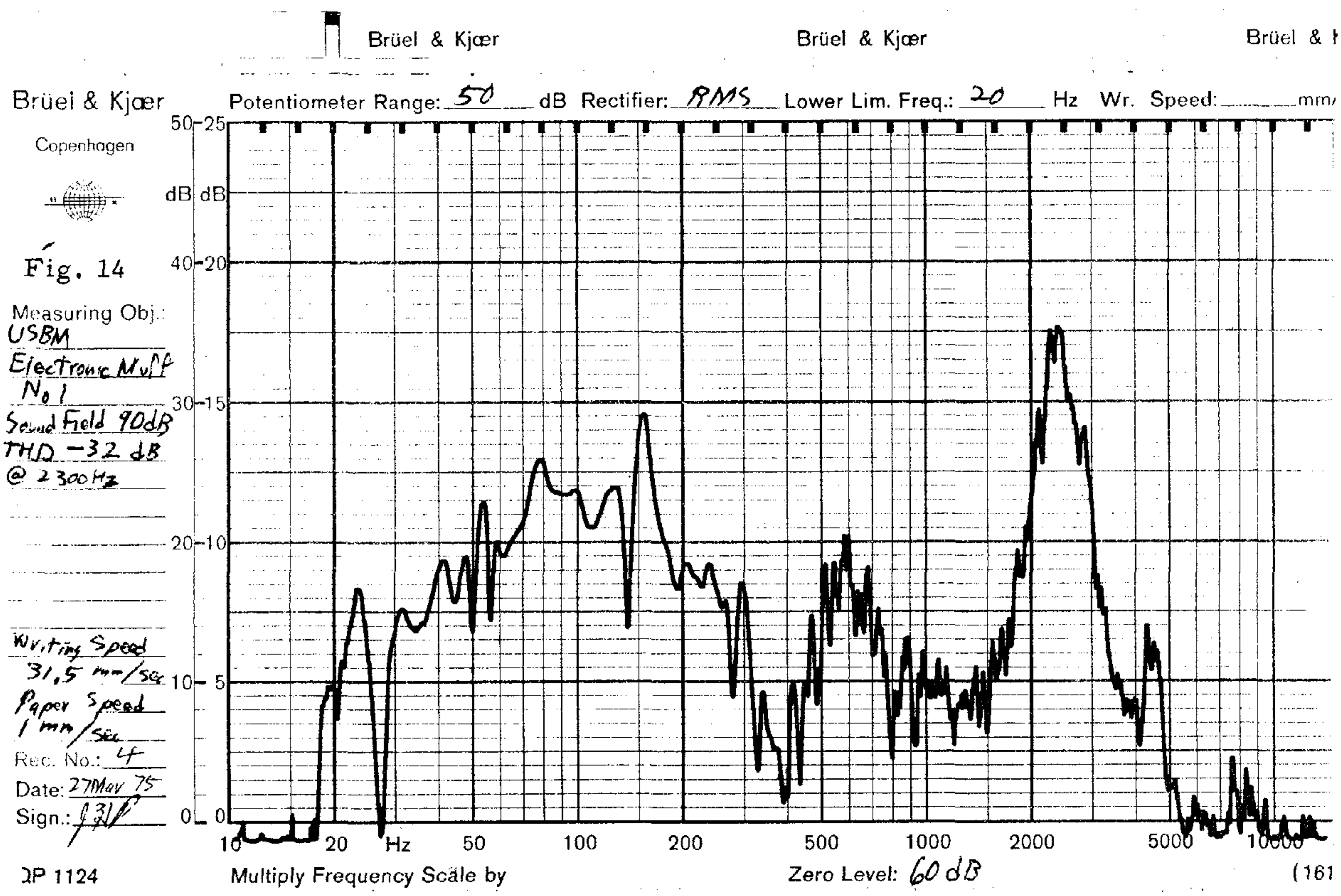
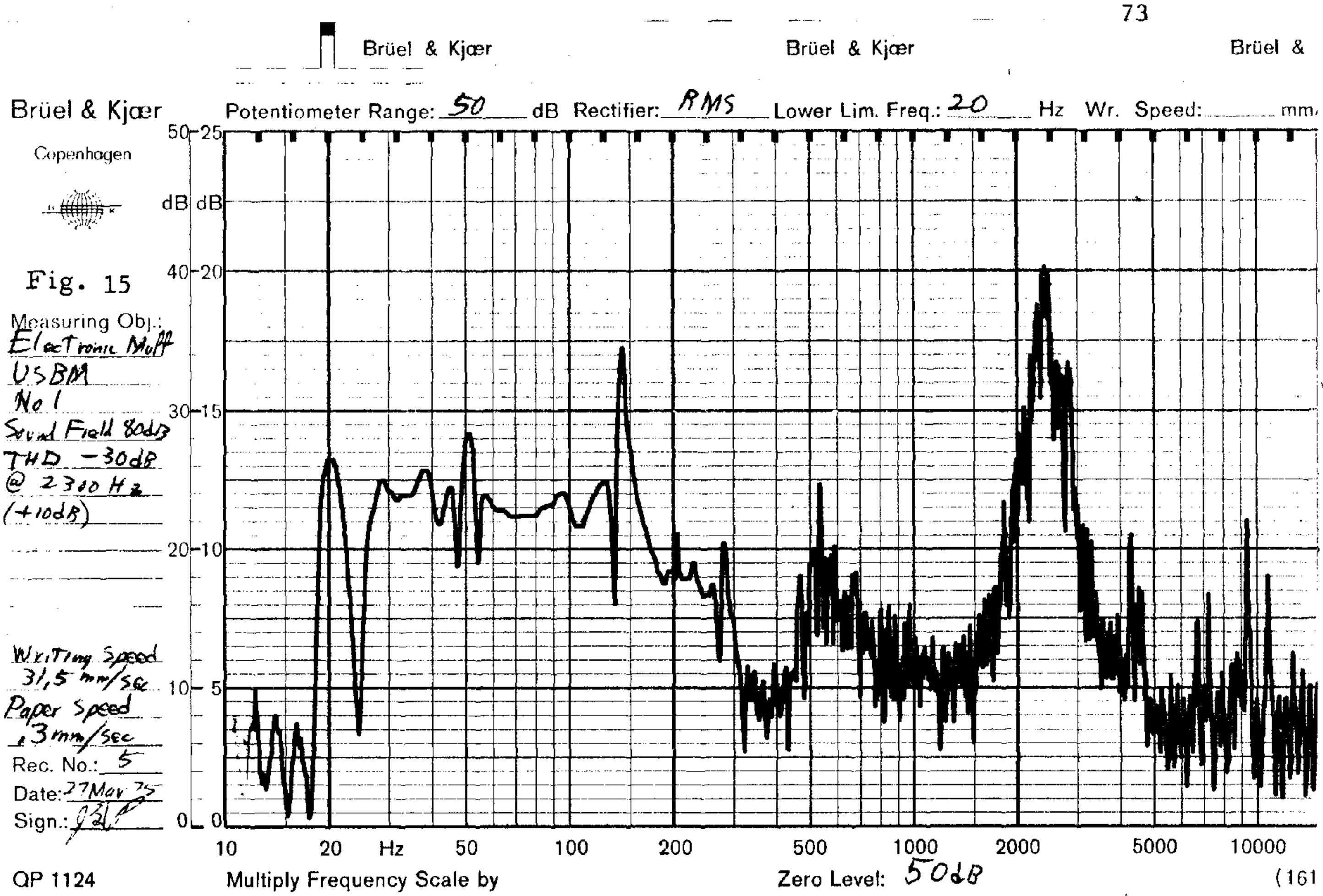
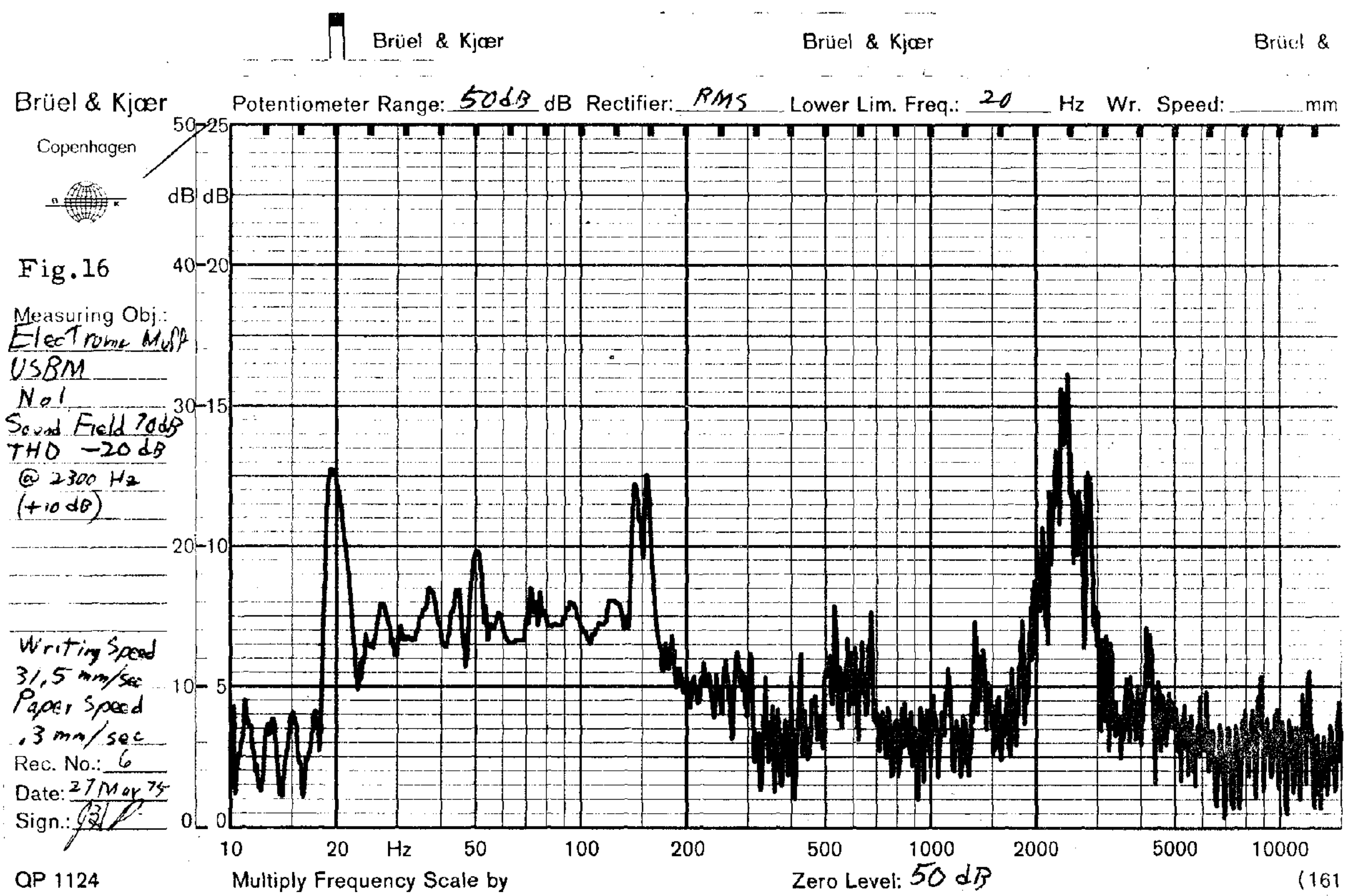

Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: mm/
Measuring Obj.: USBM Electronic Muff No 1
Sound Field 100dB
THD -36 dB
@ 2300 Hz
Writing Speed 31.5 mm/sec
Paper Speed 1 mm/sec
Rec. No.: 3
Date: 27 May 75
Sign.:
QP 1124
Multiply Frequency Scale by
Zero Level: 60 dB
10 20 Hz 50 100 200 500 1000 2000 5000 10000
50-25 dB dB 40-20 30-15 20-10 10-5 0-0
Fig. 13


Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: mm/
Measuring Obj.: USBM Electronic Muff No 1
Sound Field 90dB
THD -32 dB
@ 2300 Hz
Writing Speed 31.5 mm/sec
Paper Speed 1 mm/sec
Rec. No.: 4
Date: 27 May 75
Sign.:
QP 1124
Multiply Frequency Scale by
Zero Level: 60 dB
10 20 Hz 50 100 200 500 1000 2000 5000 10000
50-25 dB dB 40-20 30-15 20-10 10-5 0-0
Fig. 14


Brüel & Kjær
Brüel & Kjær Copenhagen
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: mm
Fig. 15
Measuring Obj.: Electronic Muff USBM No 1
Sound Field 80dB
THD -30dB @ 2300 Hz (+10dB)
Writing Speed 31,5 mm/sec
Paper Speed ,3 mm/sec
Rec. No.: 5
Date: 27 May 75
Sign.:
50 25 dB dB 40 20 30 15 20 10 10 5 0 0
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 50dB
(161


Brüel & Kjær
Brüel & Kjær Copenhagen
Potentiometer Range: 50dB dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: mm
Fig.16
Measuring Obj.: Electronic Muff USBM No 1
Sound Field 70dB
THD -20 dB @ 2300 Hz (+10 dB)
Writing Speed 31,5 mm/sec
Paper Speed ,3 mm/sec
Rec. No.: 6
Date: 27 May 75
Sign.:
50 25 dB dB 40 20 30 15 20 10 10 5 0 0
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 50 dB
(161

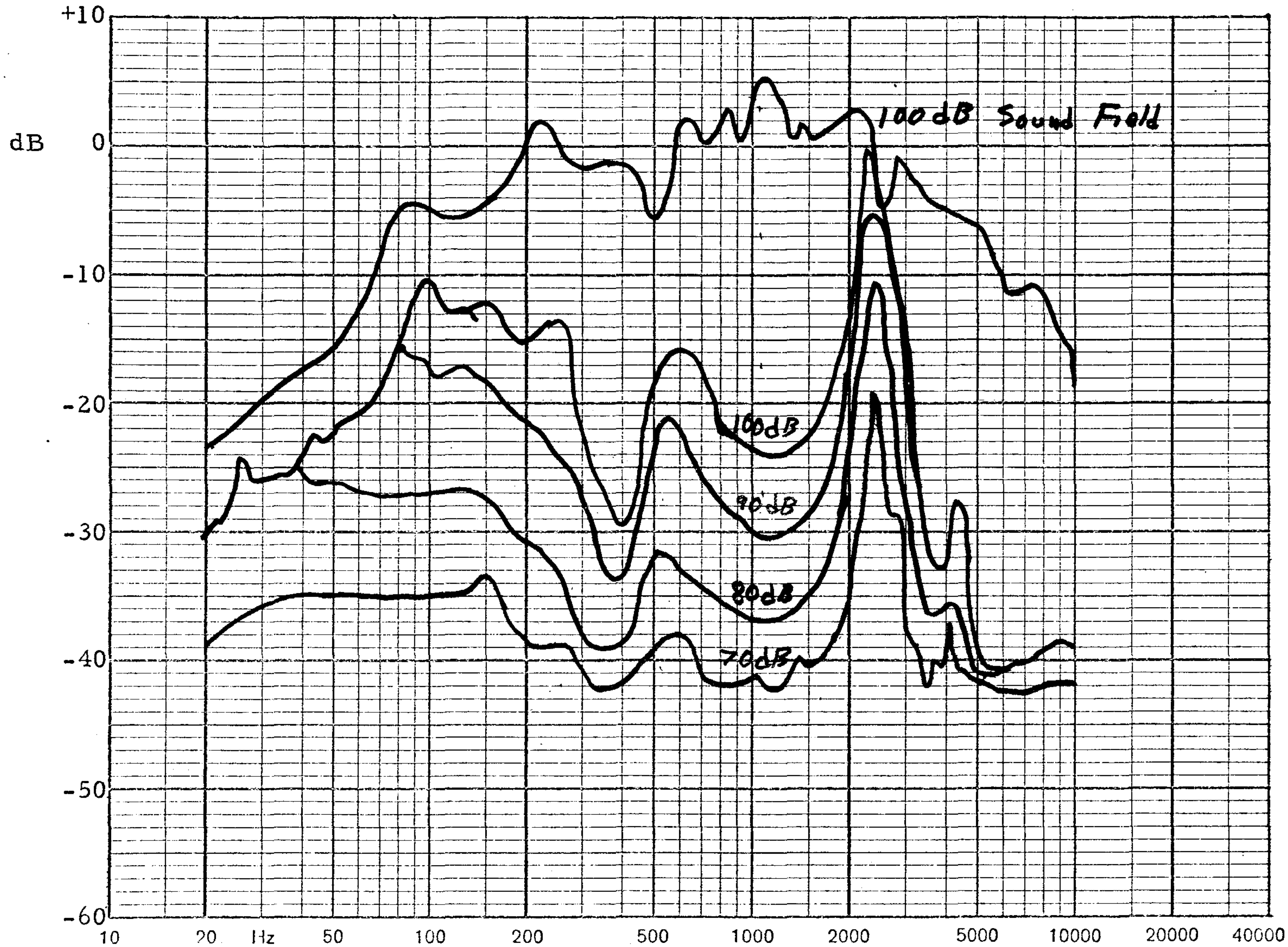
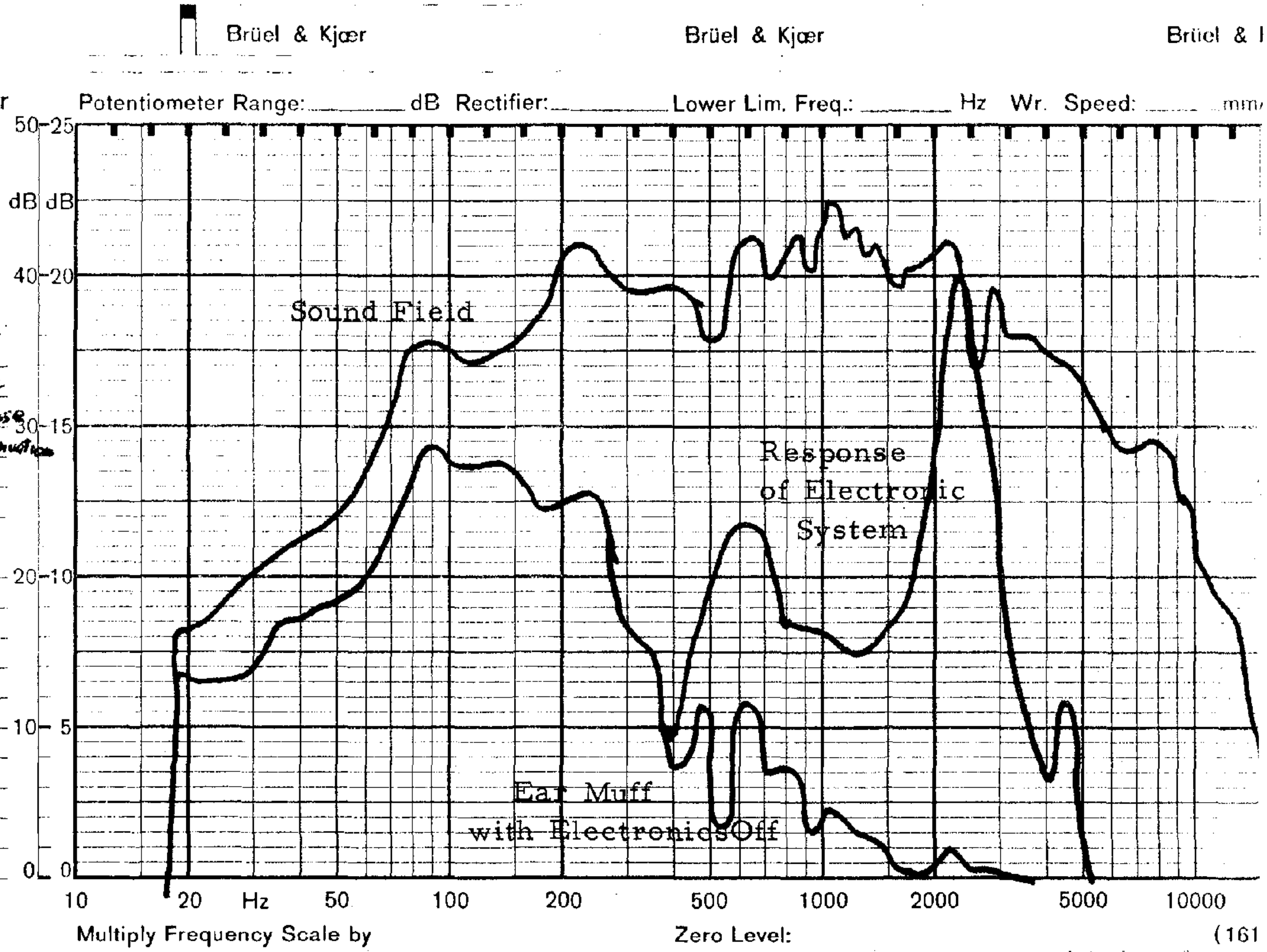


Fig. 17 Response of USBM electronic hearing protector for representative noise levels.

Fig. 18

Measuring Obj.: Contribution of Electronic System response to muff attenuation



Rec. No.:
Date:
Sign.:

QP 1124

Brüel & Kjær Brüel & Kjær Brüel &

Brüel & Kjær
Copenhagen

Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 31,5 mm

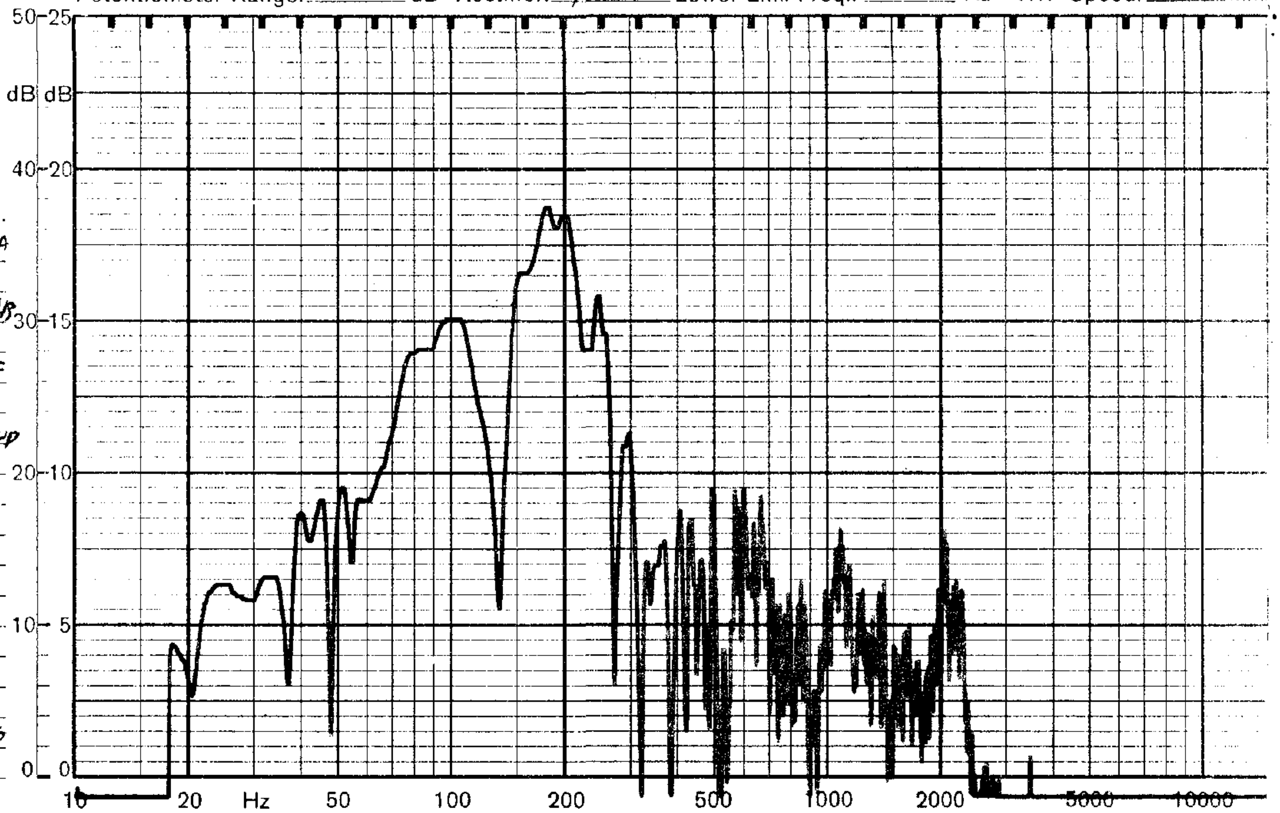
Fig. 19

Measuring Obj.: Electronic Muff
USBM No 2
Sound Field 100 dB

Electronics OFF

Signal picked up under muff

Rec. No.: 1
Date: 28 May 75
Sign.:



QP 1124 Multiply Frequency Scale by Zero Level: 60 dB (161

Brüel & Kjær Brüel & Kjær Brüel &

Brüel & Kjær
Copenhagen

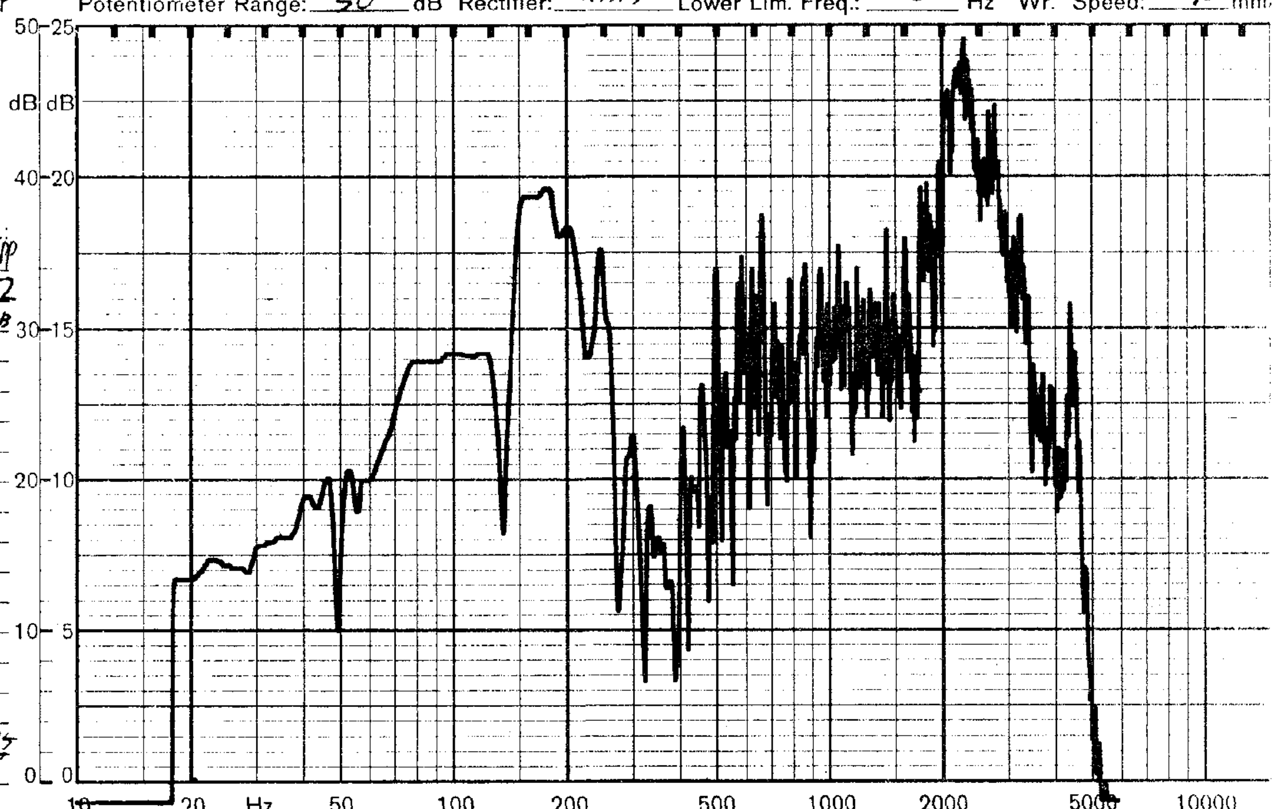
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 31,5 mm

Fig. 20

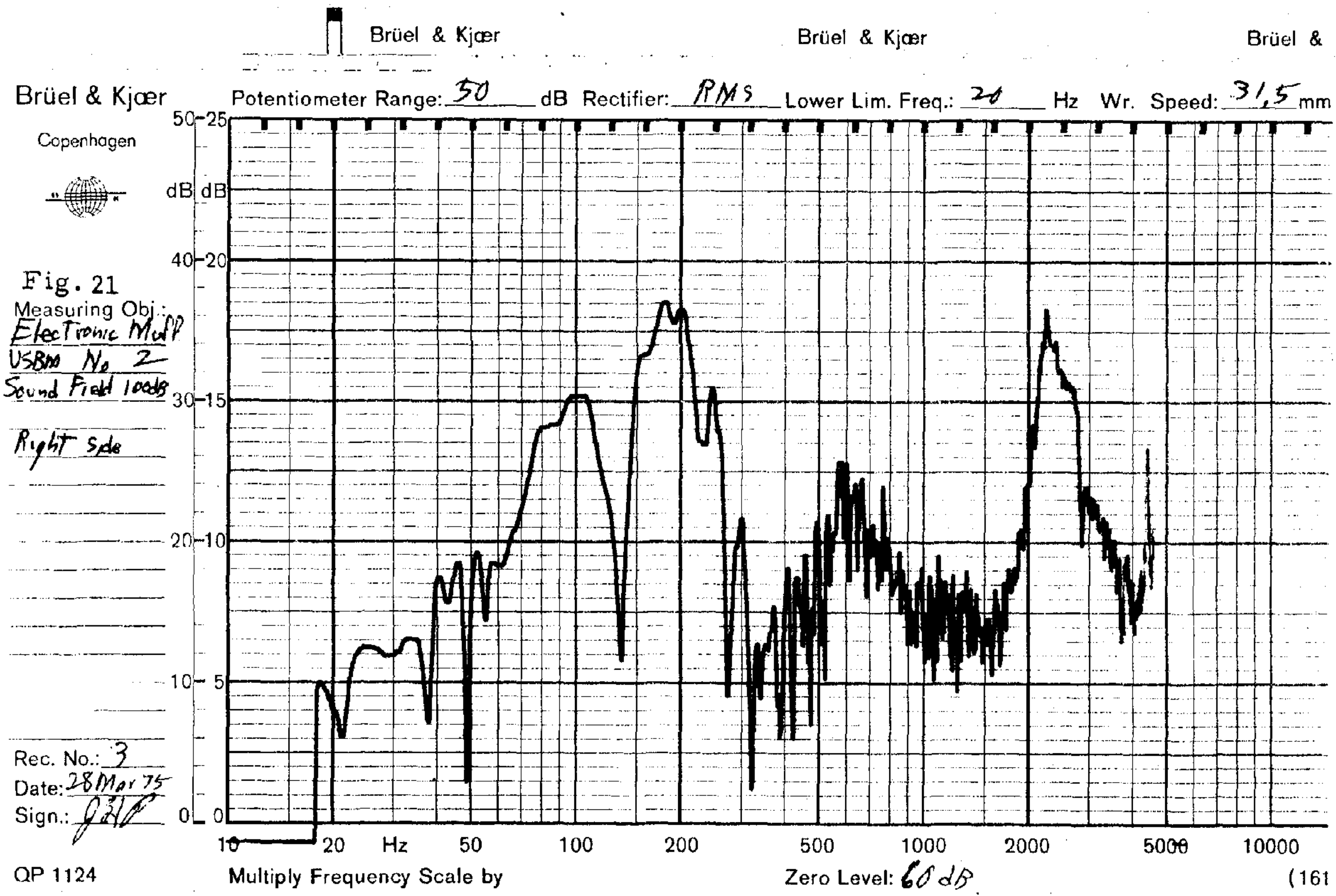
Measuring Obj.: Electronic Muff
USBM No 2
Sound Field 100 dB

Left side

Rec. No.: 2
Date: 28 May 75
Sign.:



QP 1124 Multiply Frequency Scale by Zero Level: 60 dB (161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Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 31,5 mm
Measuring Obj.: Electronic Muff USBM No 2 Sound Field 100dB
Right Side
Rec. No.: 3
Date: 28 Mar 75
Sign.: JJB
50 25
dB dB
40 20
30 15
20 10
10 5
0 0
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 60 dB

Fig. 21

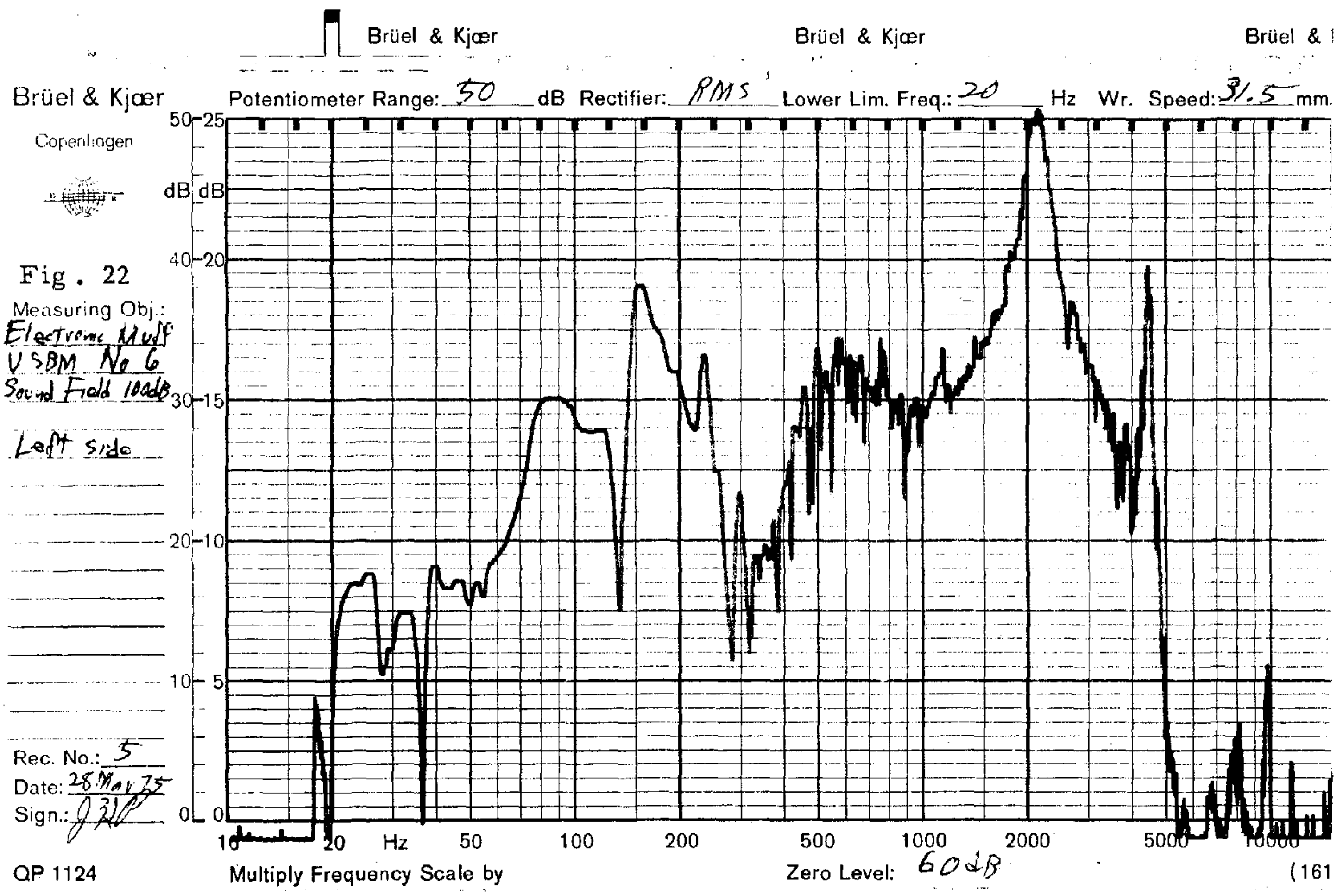
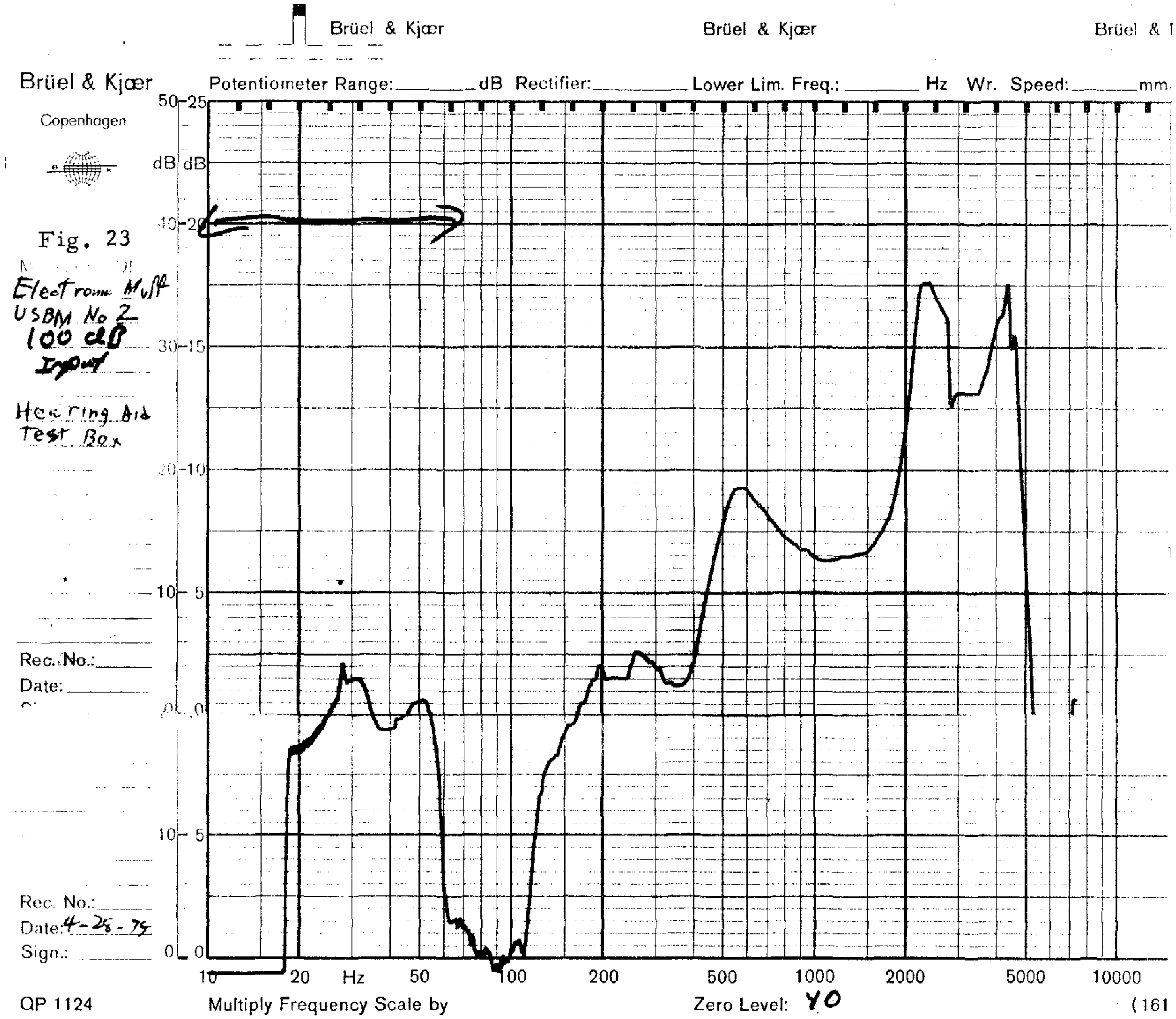
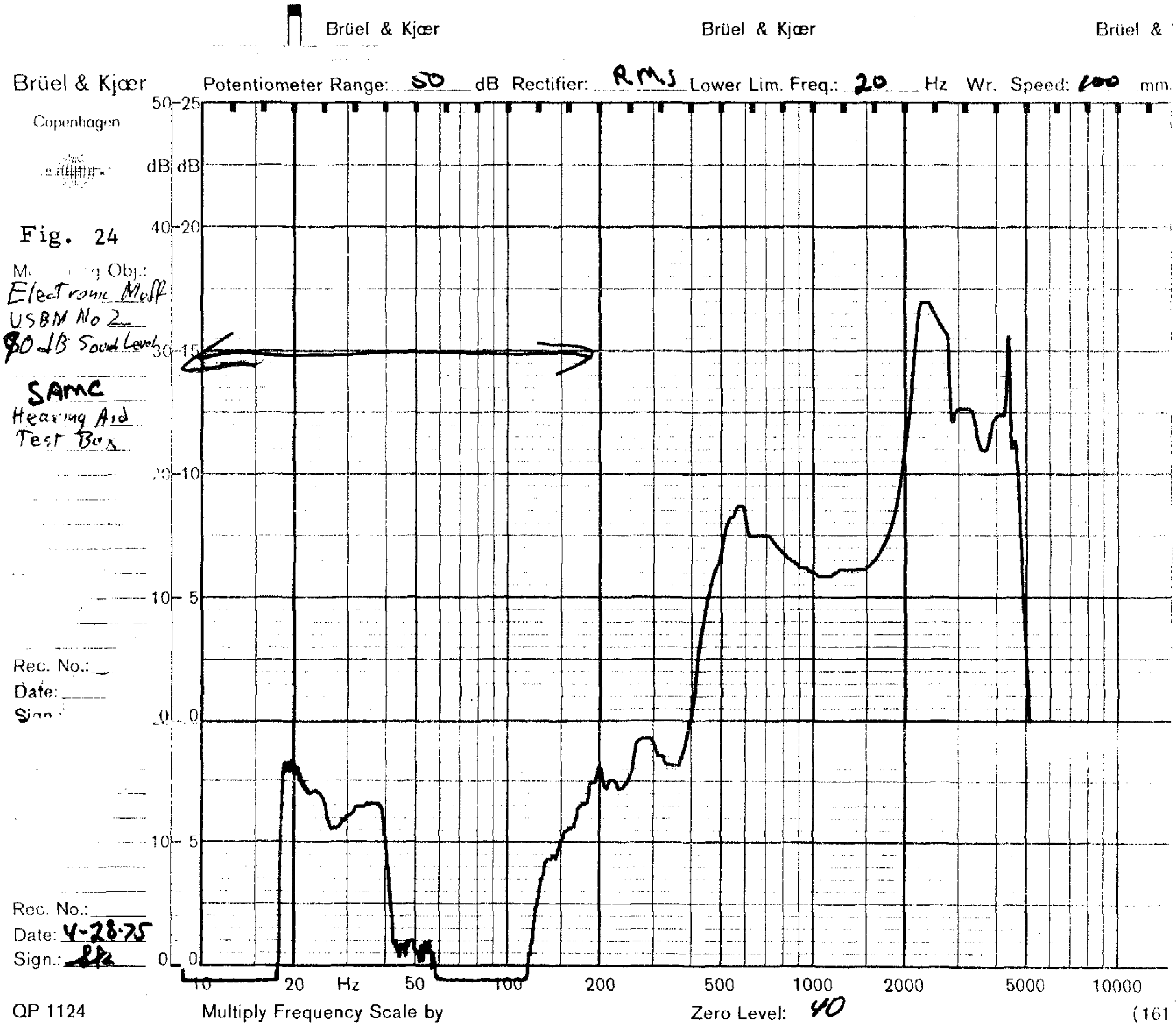
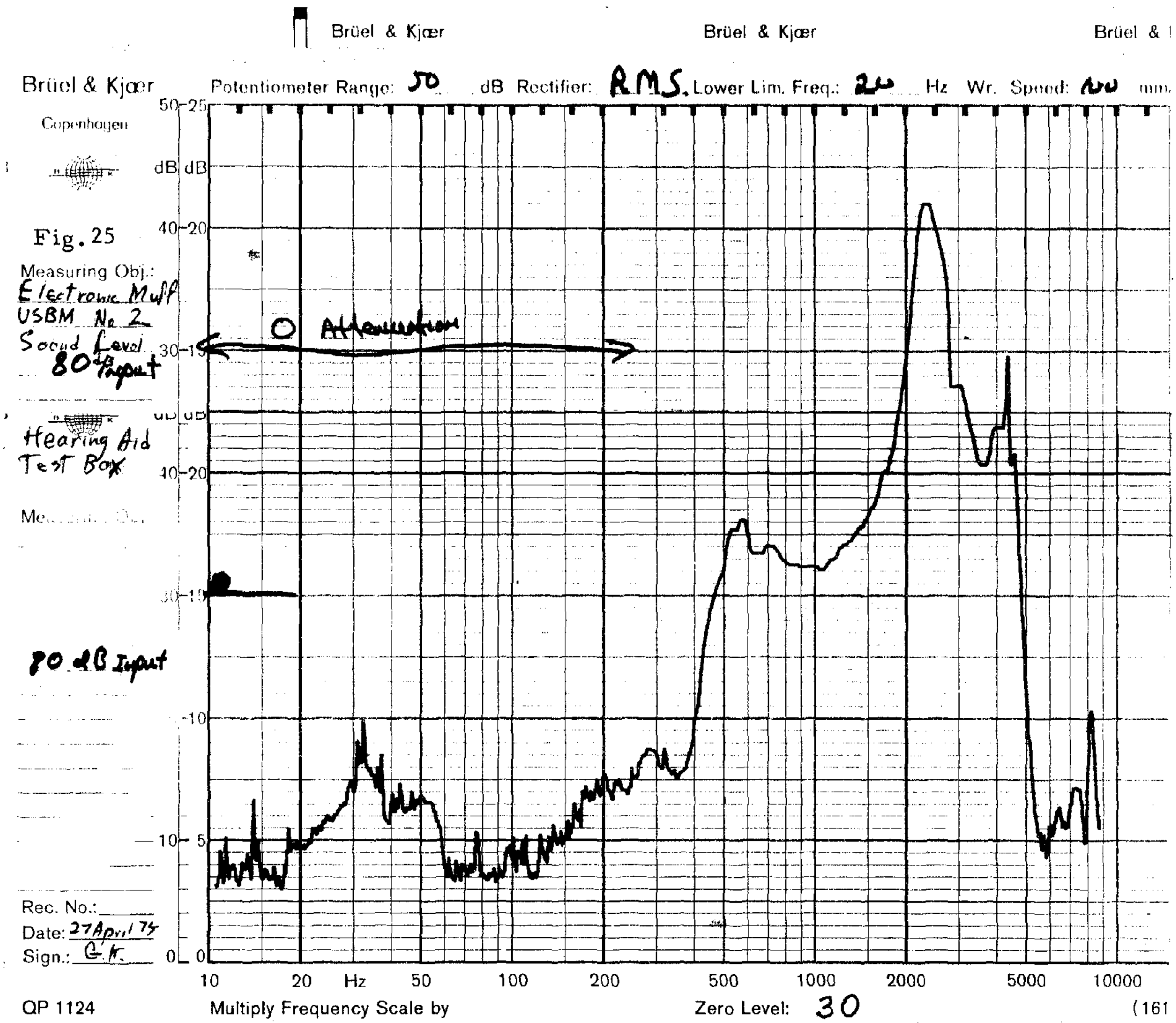

Brüel & Kjær
Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 31.5 mm.
Measuring Obj.: Electronic Muff USBM No 6 Sound Field 100dB
Left Side
Rec. No.: 5
Date: 28 Mar 75
Sign.: JJB
50 25
dB dB
40 20
30 15
20 10
10 5
0 0
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 60 dB

Fig. 22


Brüel & Kjær
Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: ____ dB Rectifier: ____ Lower Lim. Freq.: ____ Hz Wr. Speed: ____ mm
Fig. 23
Electronic Muff
USBM No 2
100 dB
Input
Hearing Aid
Test Box
Rec. No.:
Date:
Rec. No.:
Date: 4-28-75
Sign.:
50 25
dB dB
40 20
30 15
20 10
10 5
0 0
10 5
0 0
10
20
Hz
50
100
200
500
1000
2000
5000
10000
QP 1124
Multiply Frequency Scale by
Zero Level: 40
(161

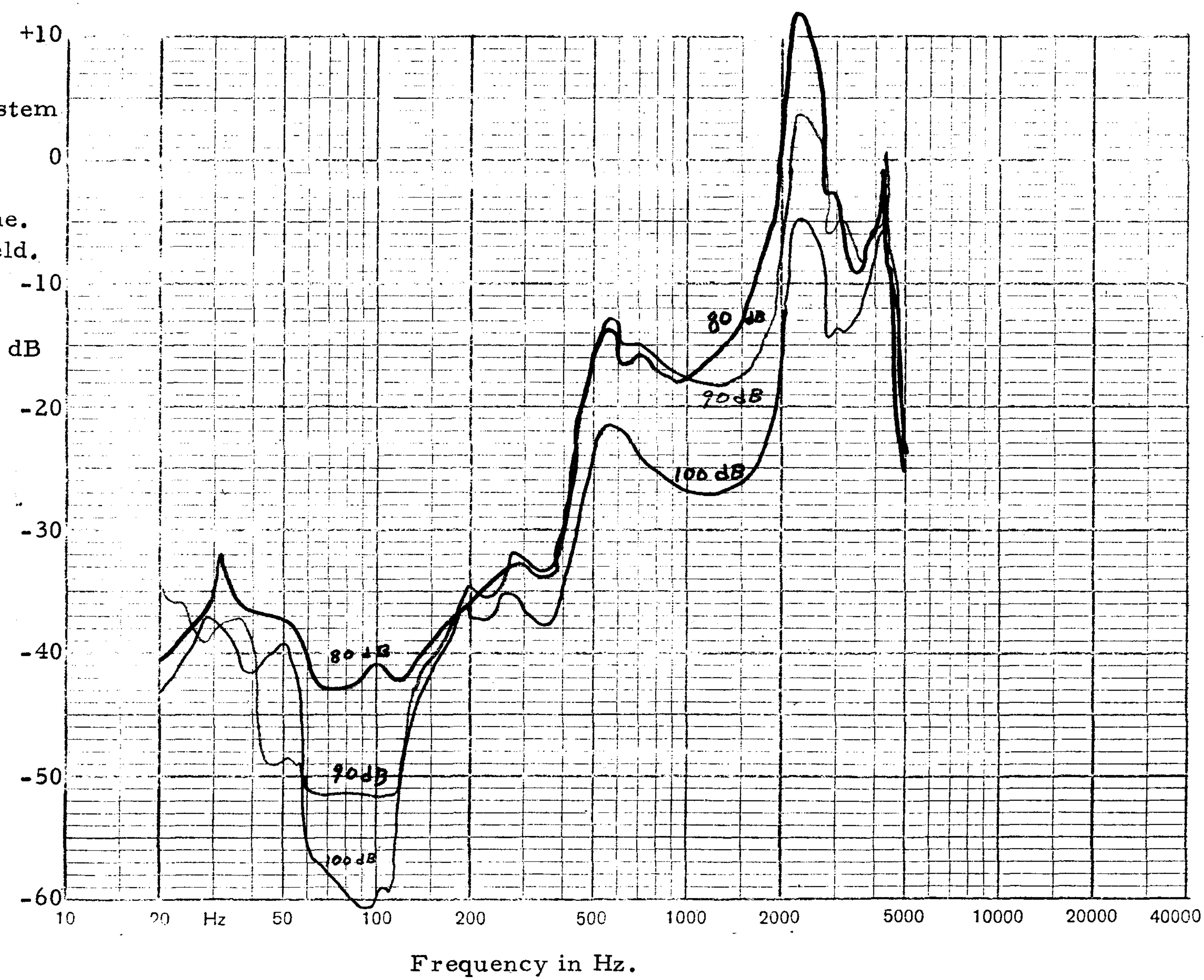

Brüel & Kjær
Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm
Fig. 24
Electronic Muff
USBM No 2
80 dB Sound Level
Same
Hearing Aid
Test Box
Rec. No.:
Date:
Sign.:
Rec. No.:
Date: 4-28-75
dB dB
50 25
40 20
30 15
20 10
10 5
0 0
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 40



Brüel & Kjær
Brüel & Kjær
Potentiometer Range: 50 dB Rectifier: RMS. Lower Lim. Freq.: 20 Hz Wr. Speed: 100
Copenhagen
Fig. 25
Measuring Obj.:
Electronic Muff
USBM No 2
Sound Level
80 dB Input
Hearing Aid
Test Box
80 dB Input
0 Attenuation
Rec. No.:
Date: 27 April 74
Sign.: G.K.
dB
50
40
30
10
0
25
20
15
5
10
20
Hz
50
100
200
500
1000
2000
5000
10000
QP 1124
Multiply Frequency Scale by
Zero Level: 30

Fig. 26

Attenuation of the MSA electronic system in sound fields of 80 dB, 90 dB, and 100 dB L_p pure tone. Earphone not in field.



Brüel & Kjær
Copenhagen

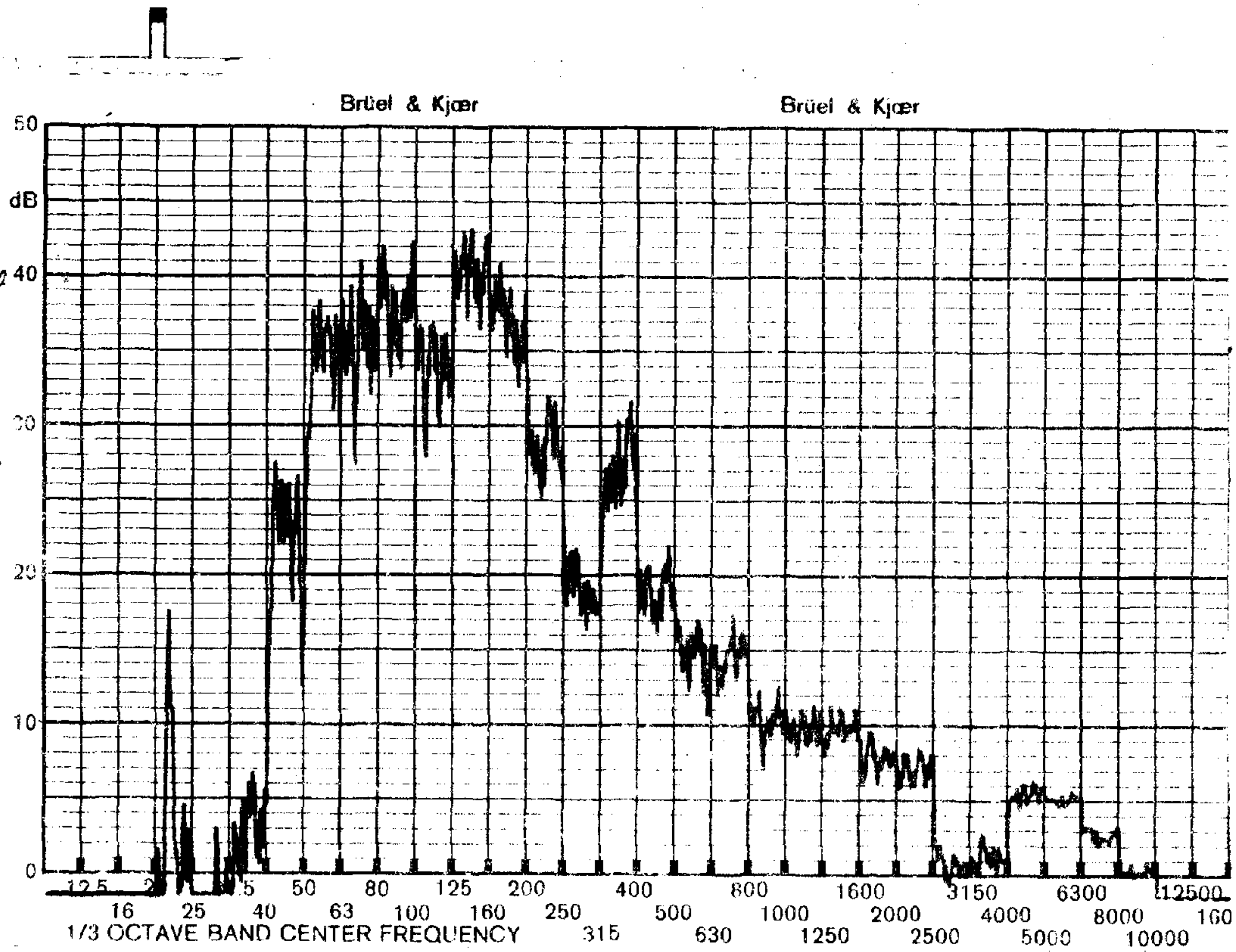
Fig. 27

Measuring Obj.: Electronic Muff USBM No 2.

100dB Diffuse Field

Electronics OFF

Rec. No.:
Date:
Sign:
Rect.:
Zero Lev.: 40
L. Lim. Fr. 20
Potm.:
Wr. Sp.:
Tape Sp.:
Multiply Freq.
Scale by:
QP 1151



Brüel & Kjær
Copenhagen

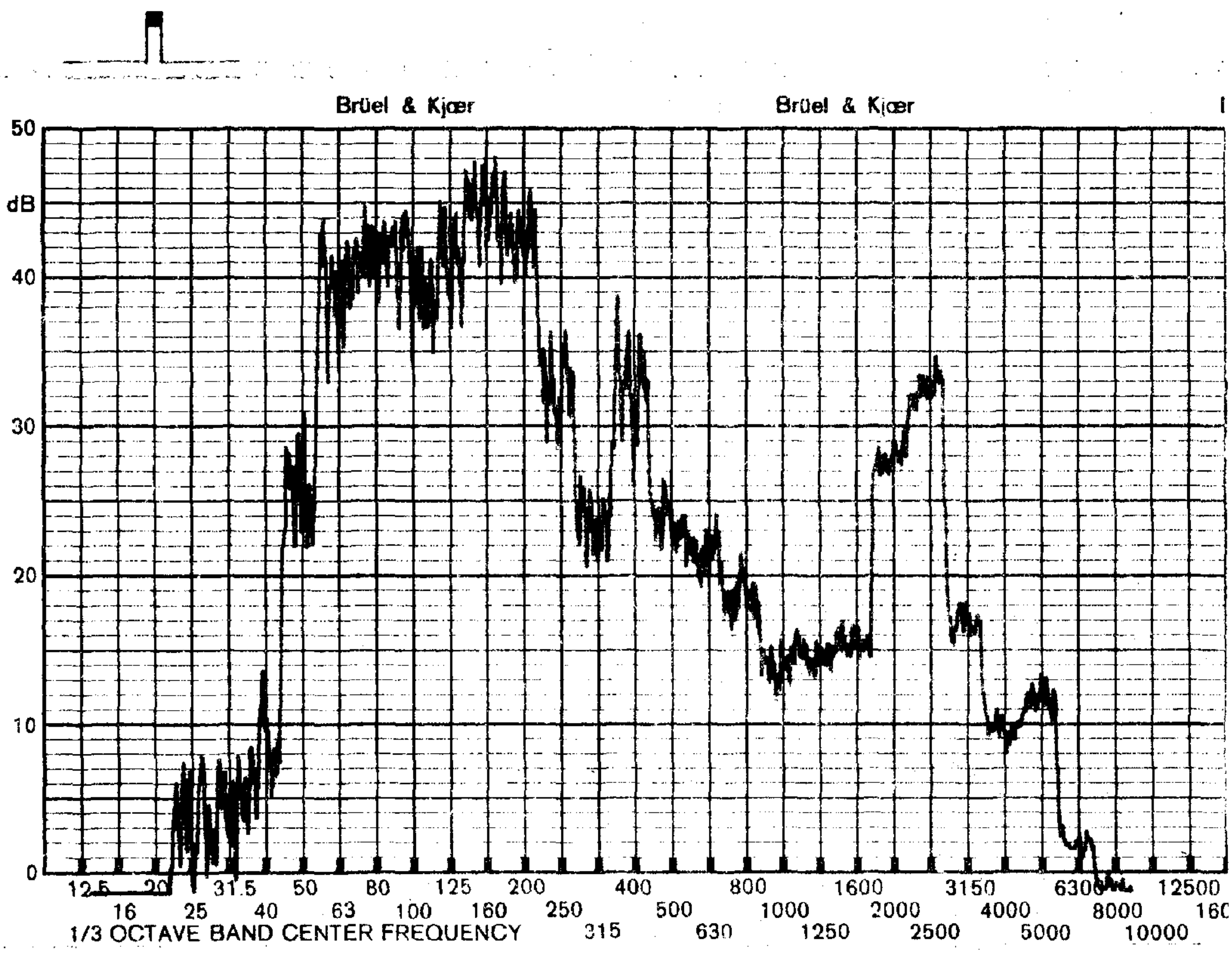
Fig. 28

Measuring Obj.: Electronic Muff USBM No 2.

115 dB Field

Electronics ON

Rec. No.:
Date:
Sign:
Rect.:
Zero Le 50
L. Lim. Fr.:
Potm.:
Wr. Sp.:
Paper Sp.:
Multiply Freq.
Scale by:
P 1151



Brüel & Kjær
Copenhagen

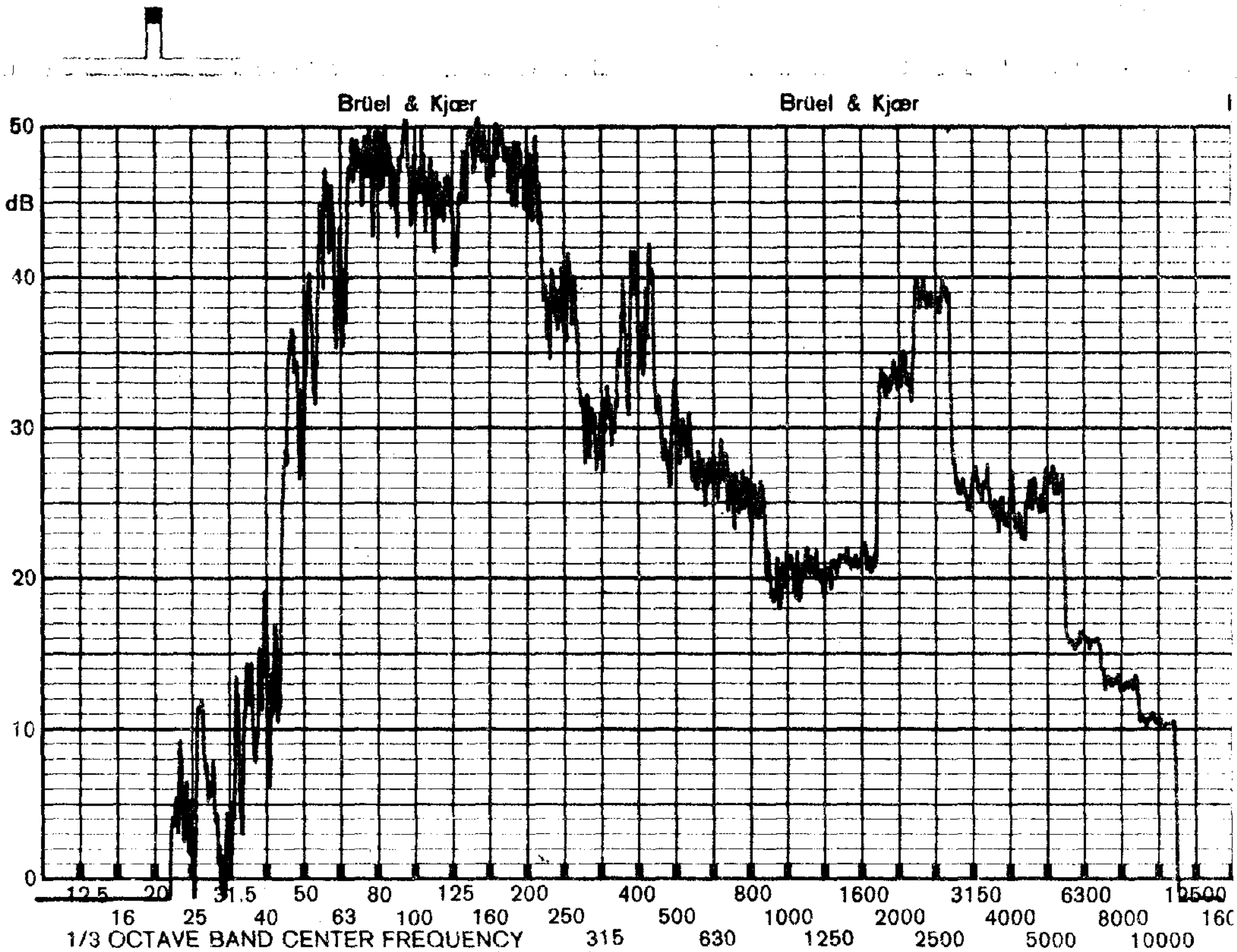
Fig.29

Measuring Obj.: Electronic Muff USBM No 2

110dB Field

Electronics ON

Rec.No.:
Date:
Sign:
Rect.:
Zero Lev.: 40
L. Lim. Fr.:
Potm.:
Wr. Sp.:
Paper Sp.:
Multiply Freq.
Scale by:
QP 1151



Brüel & Kjær
Copenhagen

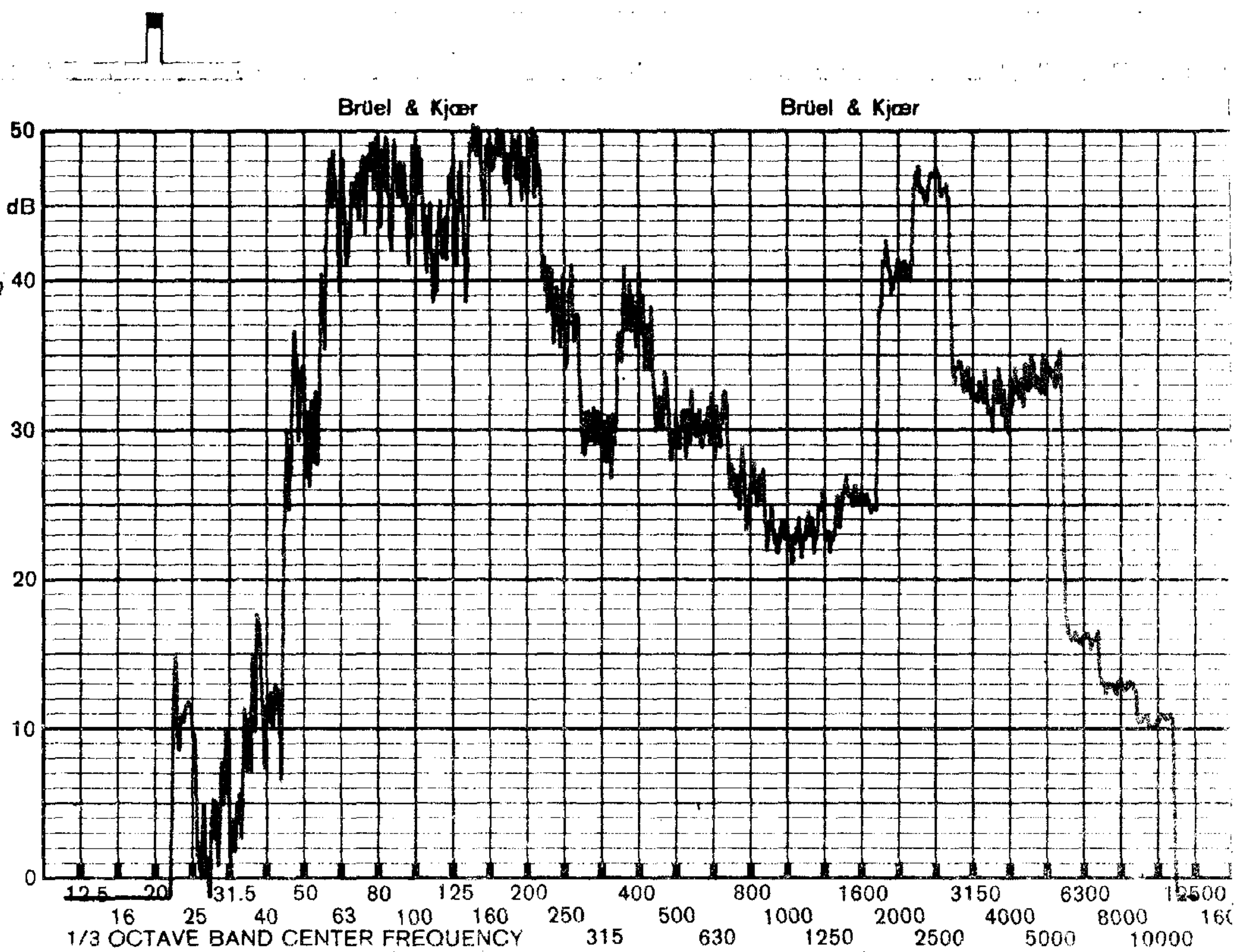
Fig. 30

Measuring Obj.: Electronic Muff USBM No 2

100dB Field

Electronics ON

Rec.No.:
Date:
Sign:
Rect.:
Zero Lev.: 30
L. Lim. Fr.: 20
Potm.:
Wr. Sp.:
Paper Sp.:
Multiply Freq.
Scale by:
QP1151



Brüel & Kjær
Copenhagen

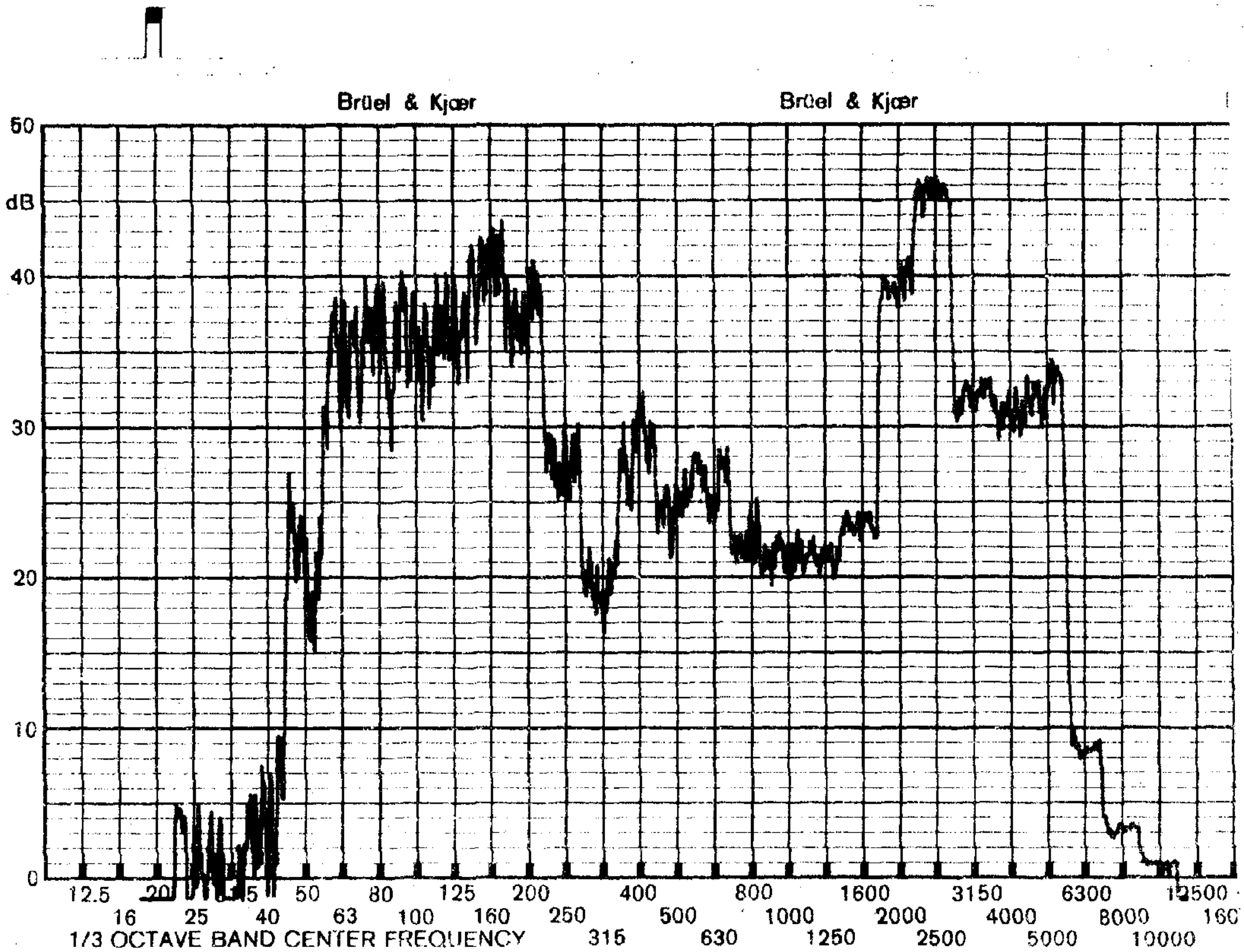
Fig. 31

Measuring Obj.: Electronic Muff USBM No 2

90 dB Field

Electronics ON

Rec.No.:
Date:
Sign:
Rect.:
Zero Lev. 30
L. Lim. Fr.:
Potm.:
Wr. Sp.:
Paper Sp.:
Multiply Freq.
Scale by:
QP 1151



Brüel & Kjær
Copenhagen

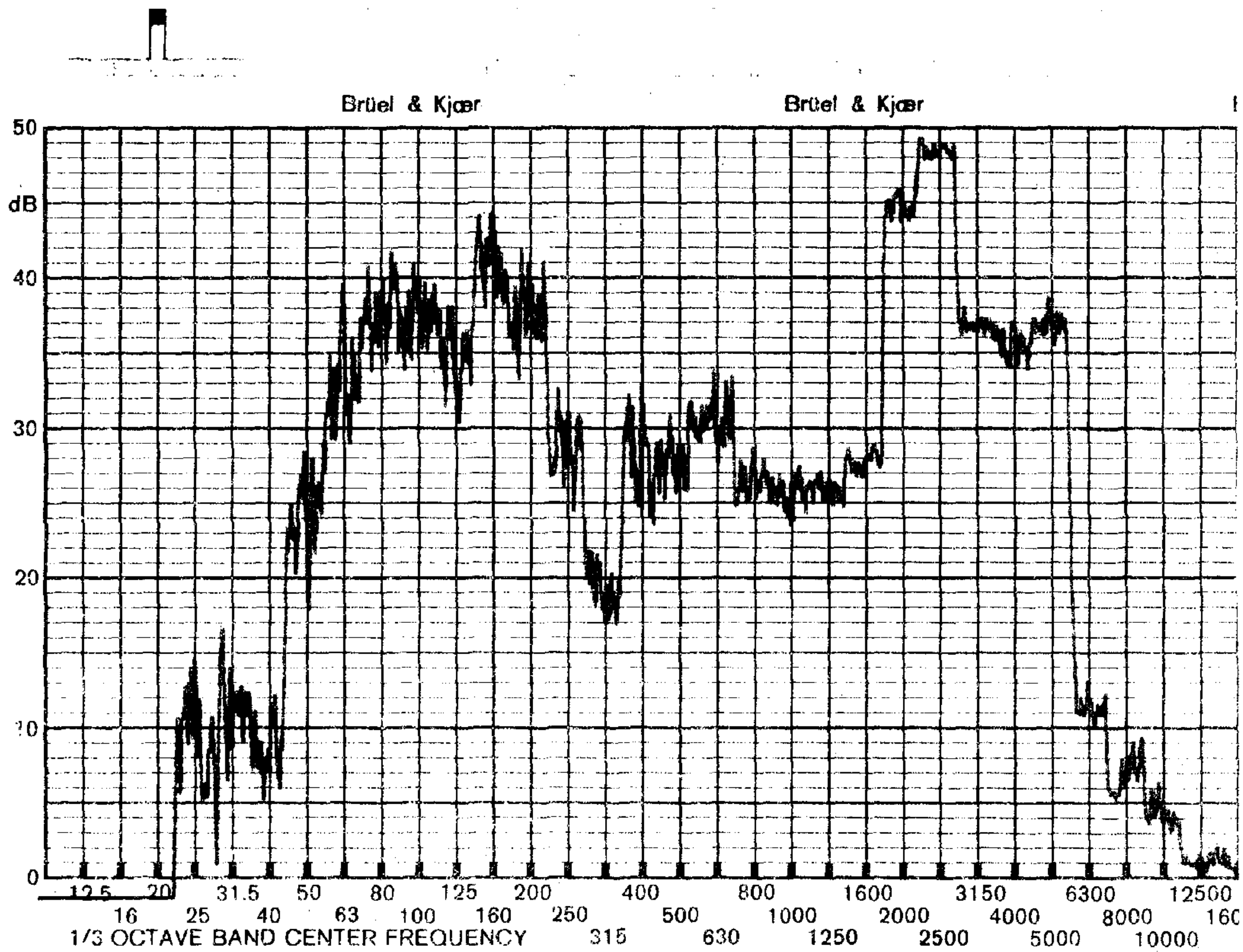
Fig. 32

Measuring Obj.: Electronic Muff USBM No 2

80 dB Field

Electronics ON

Rec.No.:
Date:
Sign:
Rect.:
Zero Lev 20
L. Lim. Fr. 20
Potm.:
Wr. Sp.:
Paper Sp.:
Multiply Freq.
Scale by:
QP 1151



Brüel & Kjær
Copenhagen

Fig. 33

Measuring Obj.: Electronic Muff
USBM No 2

70 dB Field

Electronics ON

Rec.No
Date:
Sign:
Rect.:
Zero Lev.: 20
L. Lim. Fr.: 20
Potm.:
Wr. Sp.:
Paper Sp.:
Multiply Freq.
Scale by:
QP 1151

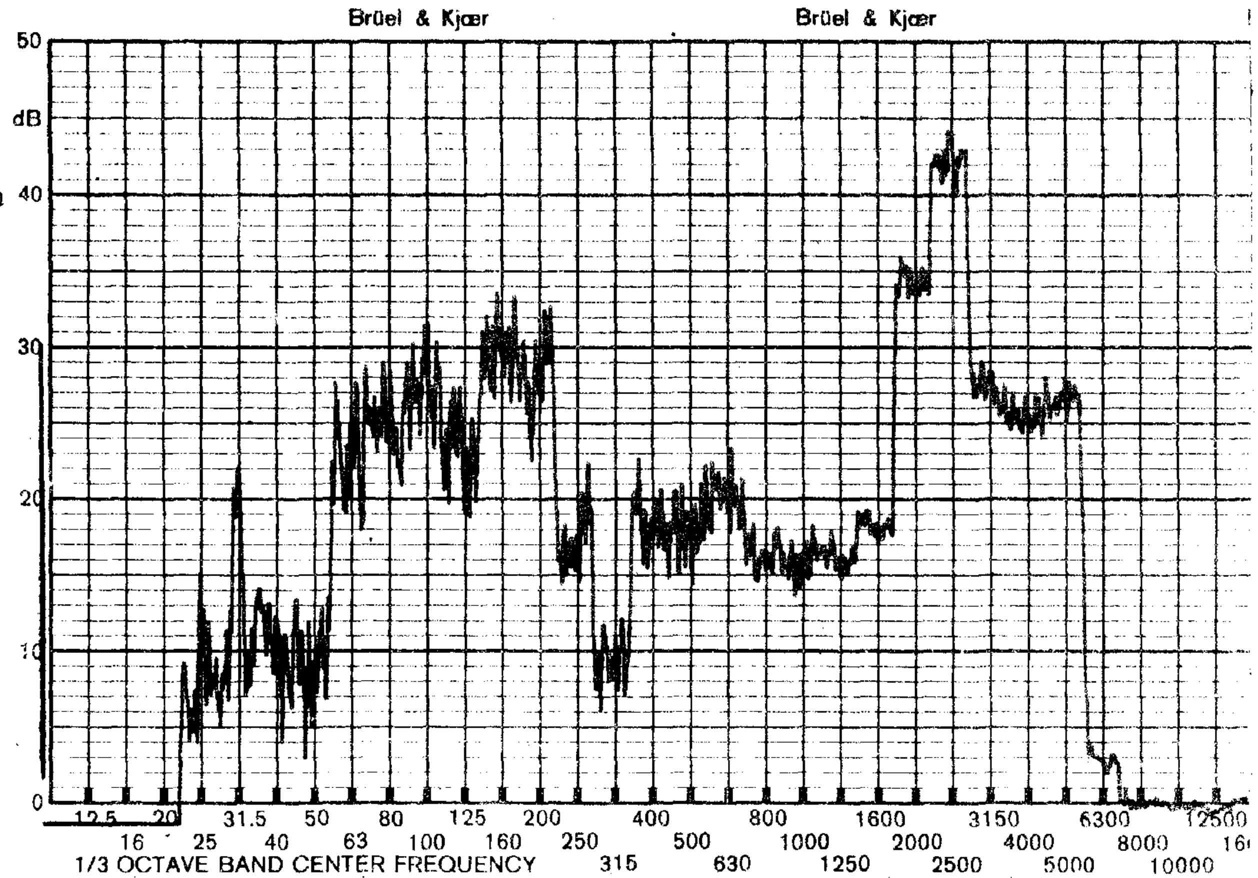
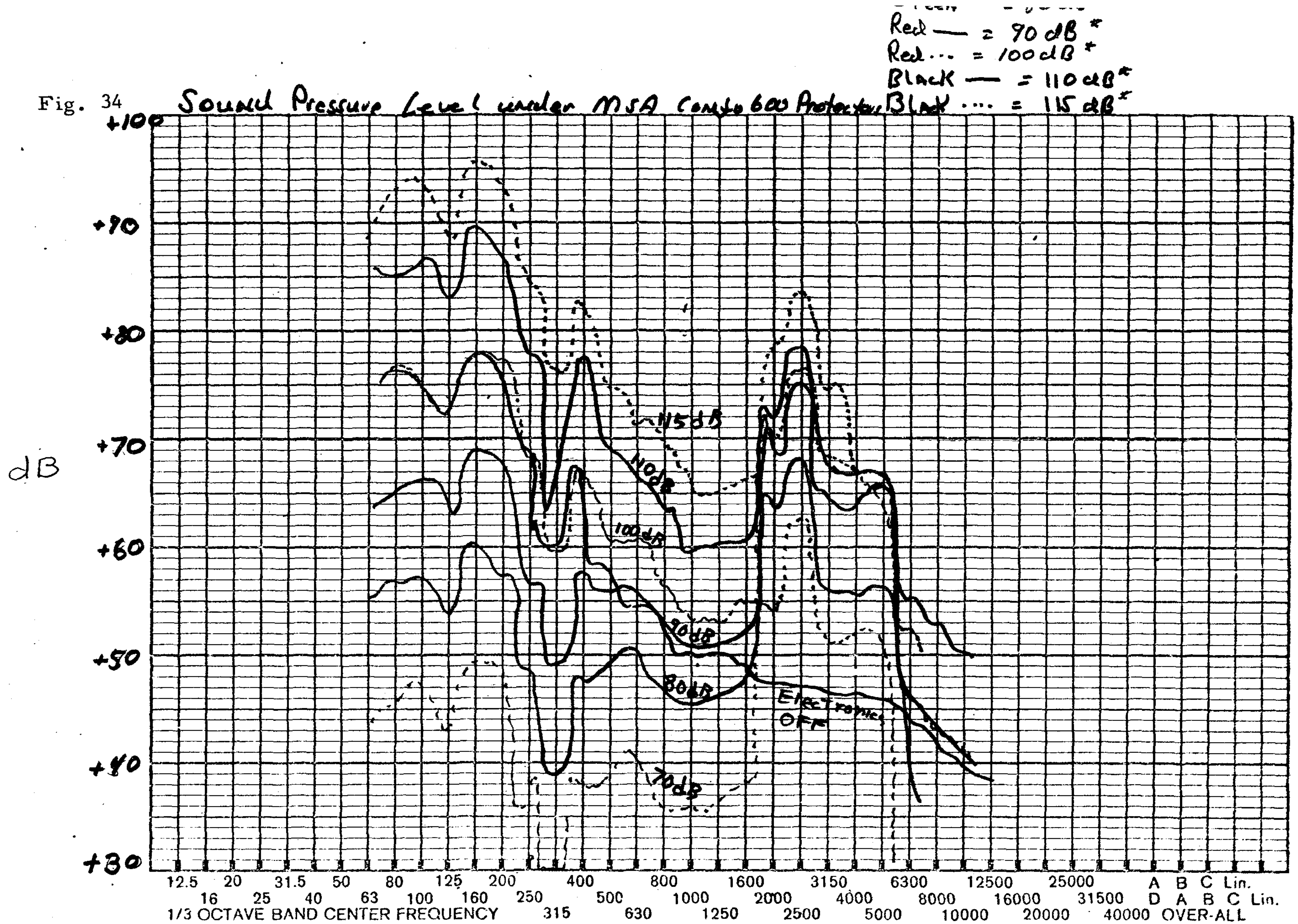
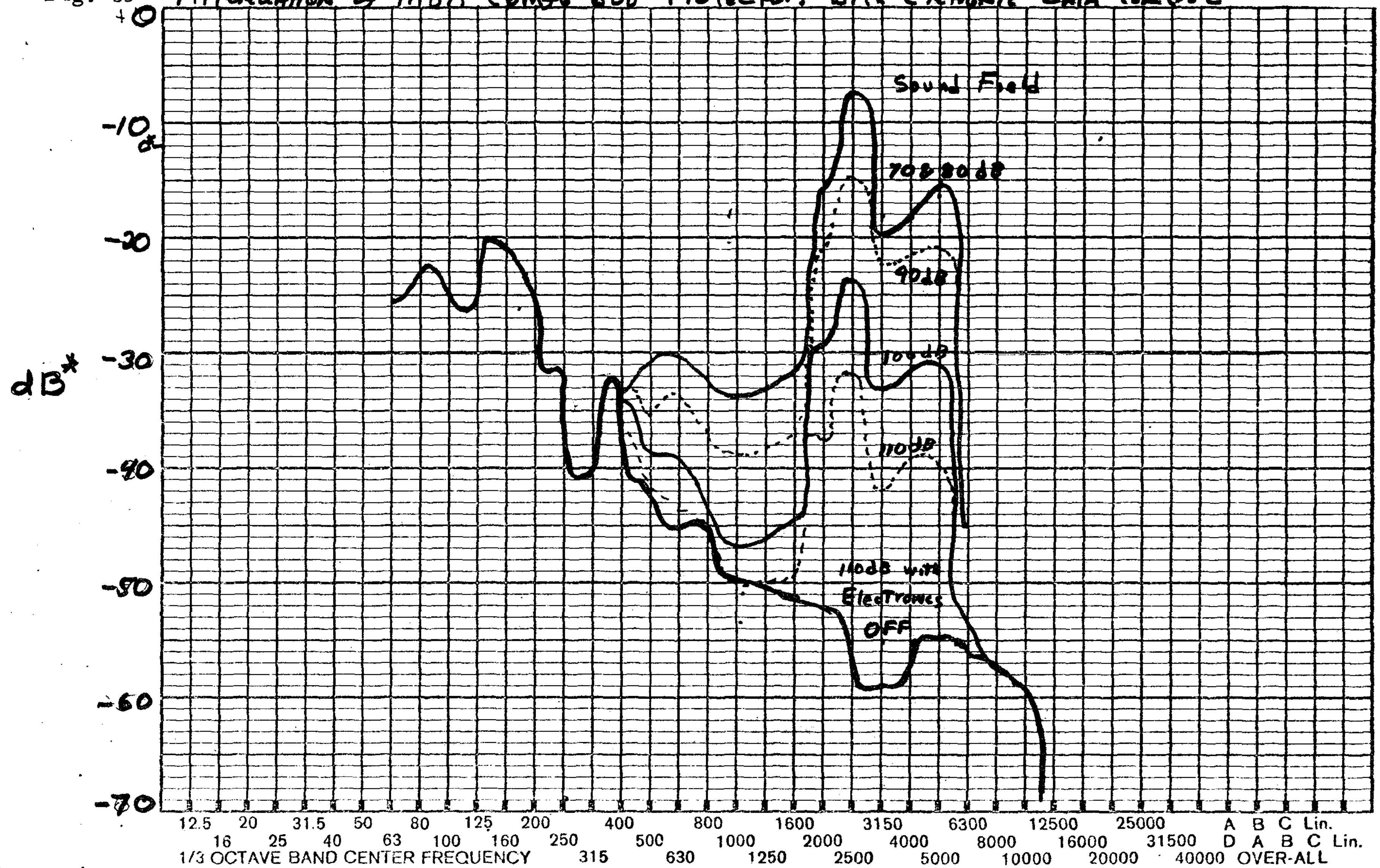


Fig. 34 Sound Pressure Level under MSA Comfo 600 Protector



* Sound field measured in dB per 63Hz ~~to 8000Hz~~.
for dB per 1/3 octave subtract 12 dB

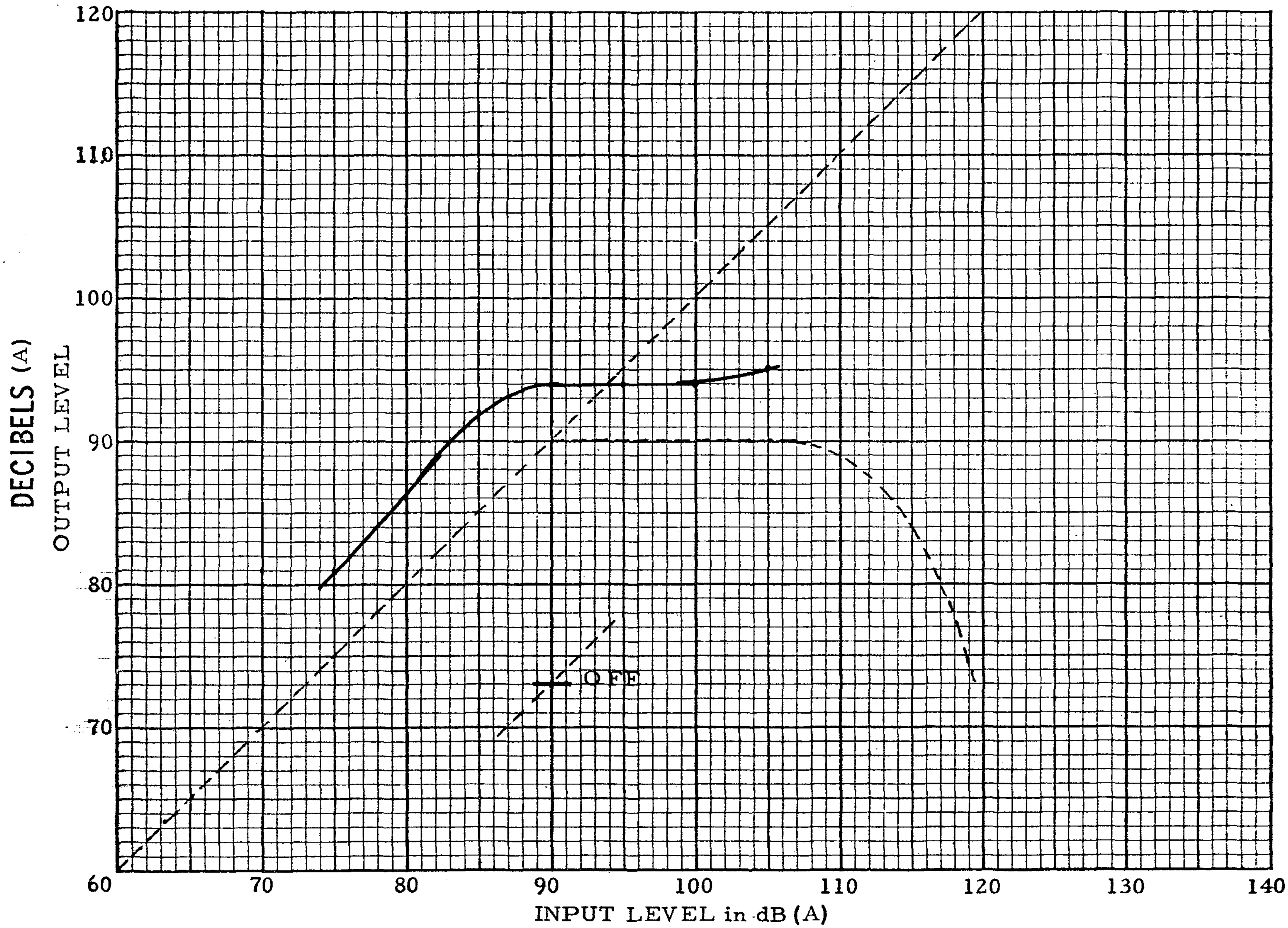
Fig. 35 Attenuation of MSA Comfo 600 Protectors with Electronic Gain Control



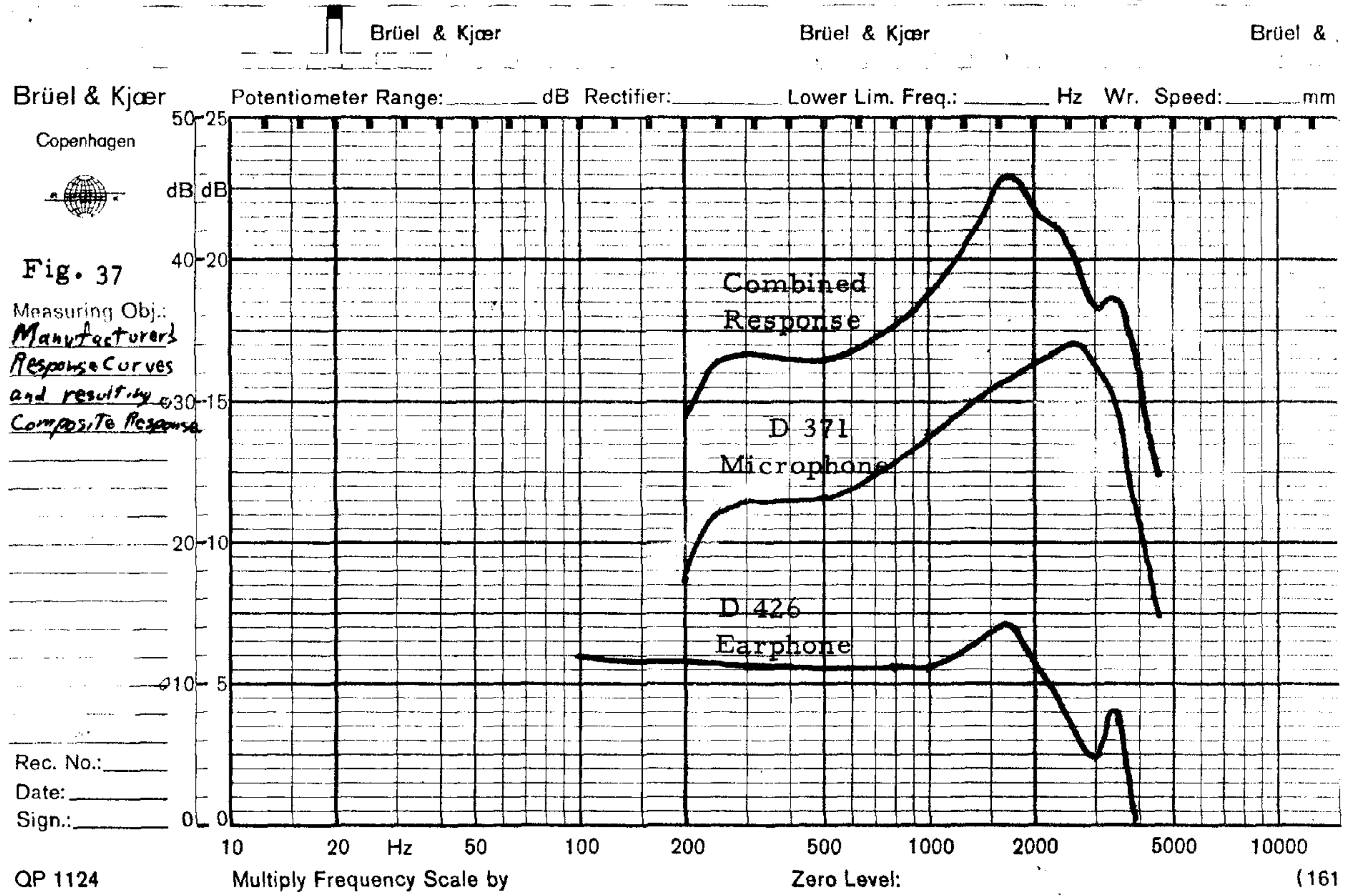
* Attenuation measured in dB per 63 Hz to 8000 Hz

Add +12 dB to get Attenuation per 1/3 octave

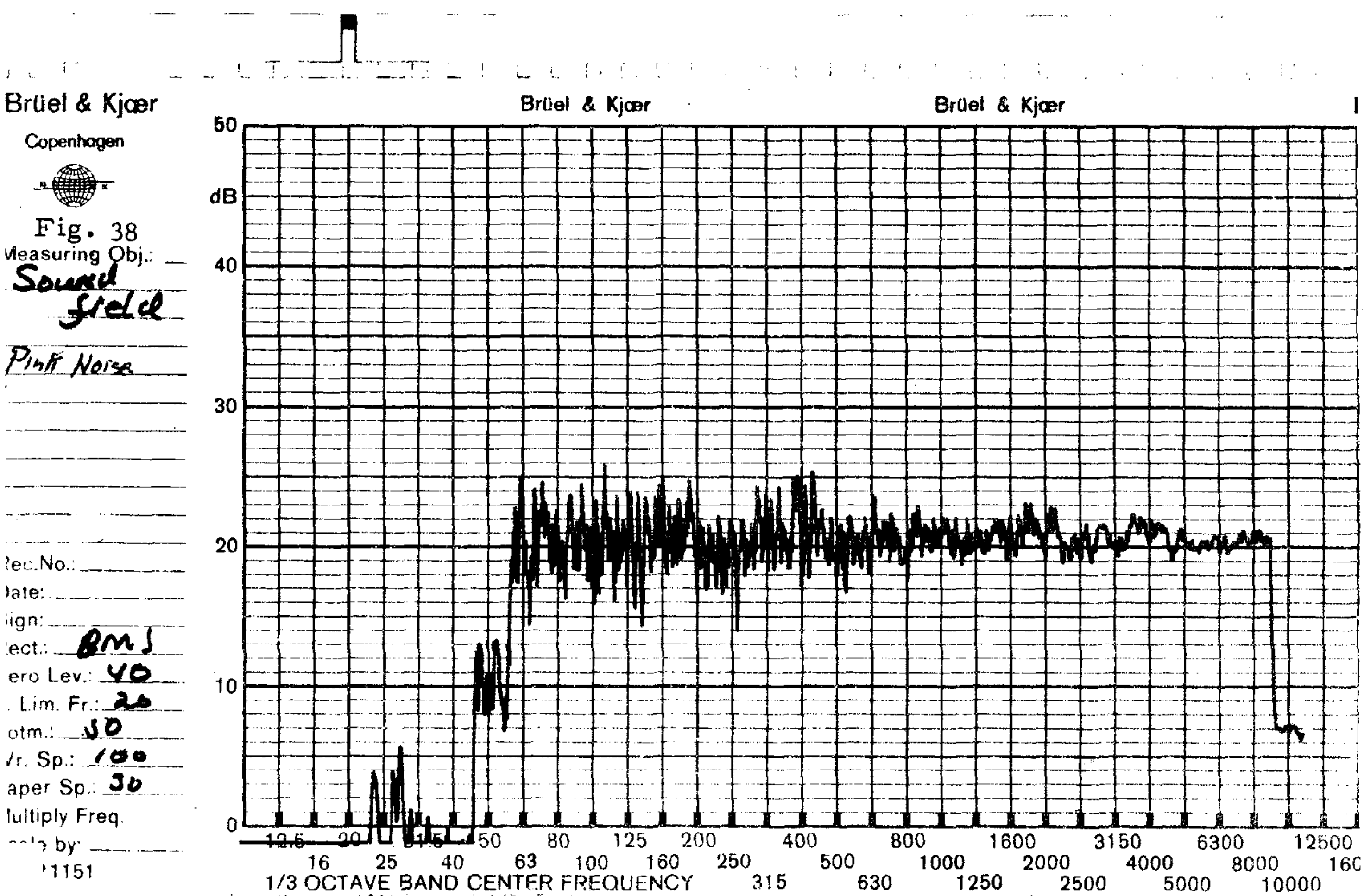
FIGURE 36



TRANSFER CHARACTERISTIC OF USBM - MSA ELECTRONIC PROTECTOR No. 6



Brüel & Kjær
Copenhagen
Fig. 37
Measuring Obj.:
Manufacturers Response Curves and resulting Composite Response
Potentiometer Range: dB Rectifier: Lower Lim. Freq.: Hz Wr. Speed: mm
Combined Response
D 371 Microphone
D 426 Earphone
Rec. No.:
Date:
Sign.:
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level:



Brüel & Kjær
Copenhagen
Fig. 38
Measuring Obj.:
Sound field
Pink Noise
Rec.No.:
Date:
Sign.:
Rect.: RMS
Zero Lev.: 40
Lim. Fr.: 20
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
1/3 OCTAVE BAND CENTER FREQUENCY

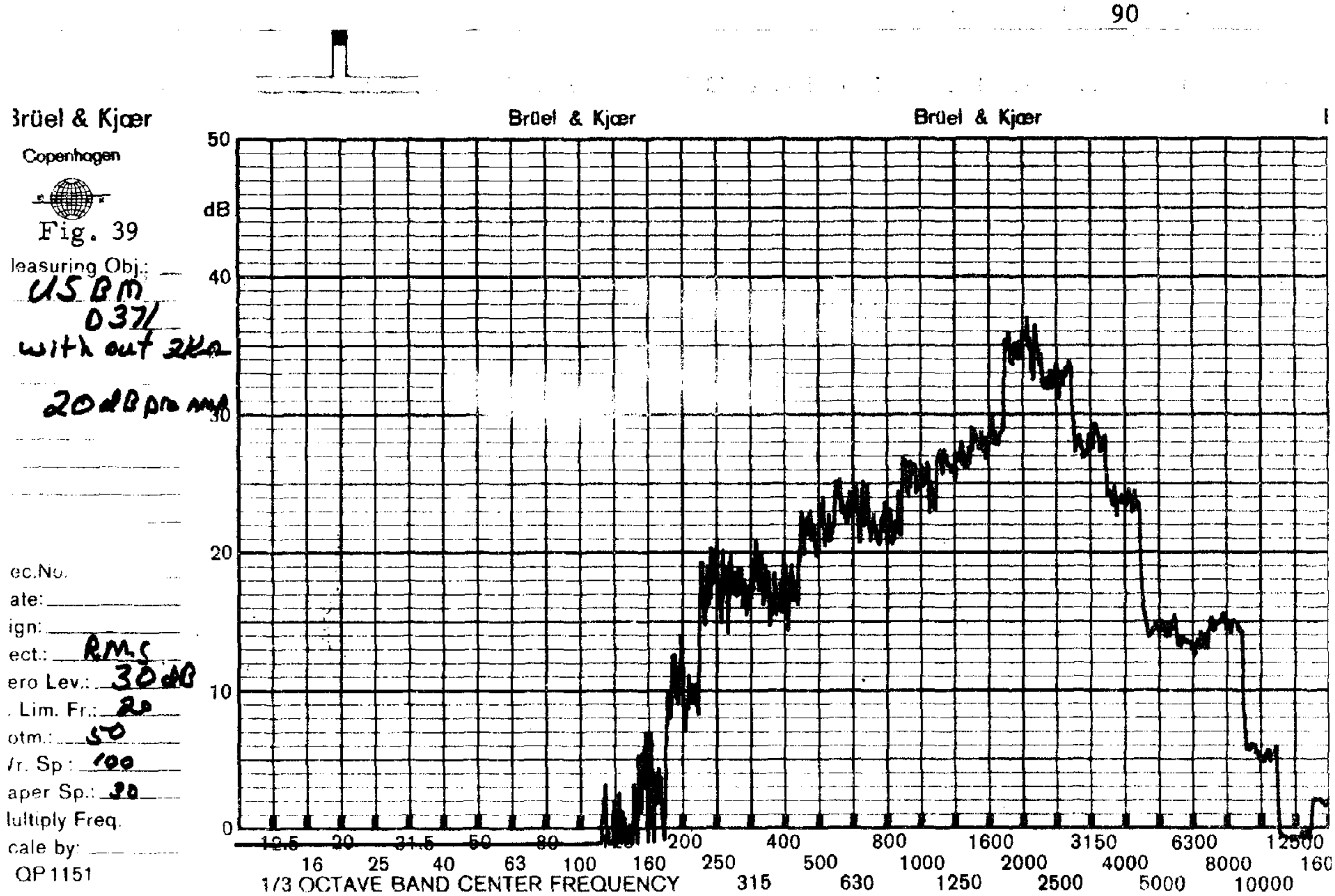

Brüel & Kjær
Copenhagen
Measuring Obj.: USBM D371 with out 2kΩ 20dB pre amp
Rec.No.
Date:
Sign:
Rect.: RMS
Zero Lev.: 30 dB
L. Lim. Fr.: 20
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq. Scale by:
QP 1151
Brüel & Kjær
Brüel & Kjær
50
dB
40
30
20
10
0
12.5 20 31.5 50 80 125 200 400 800 1600 3150 6300 12500
16 25 40 63 100 160 250 500 1000 2000 4000 8000 160
1/3 OCTAVE BAND CENTER FREQUENCY 315 630 1250 2500 5000 10000

Fig. 39

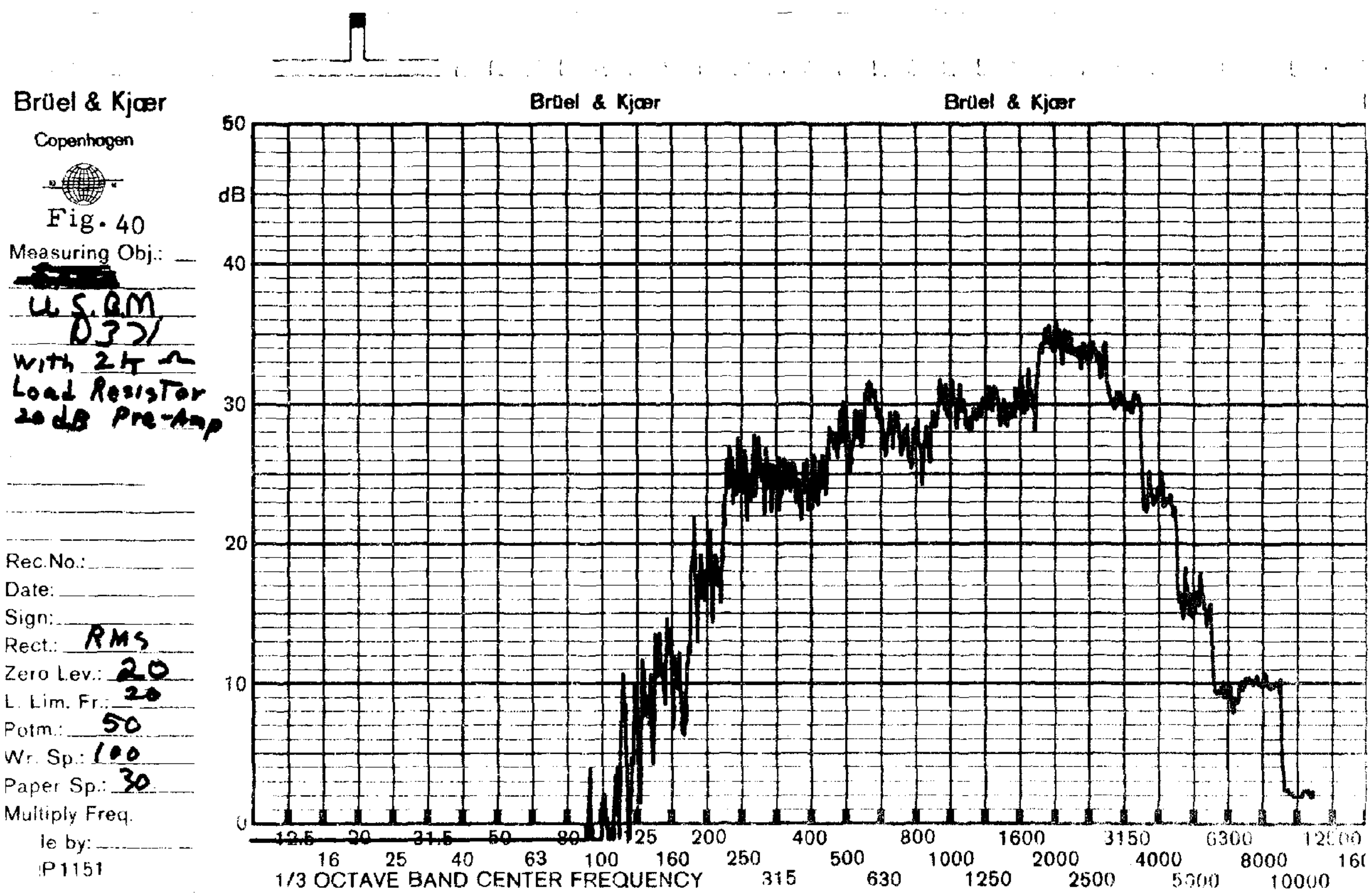
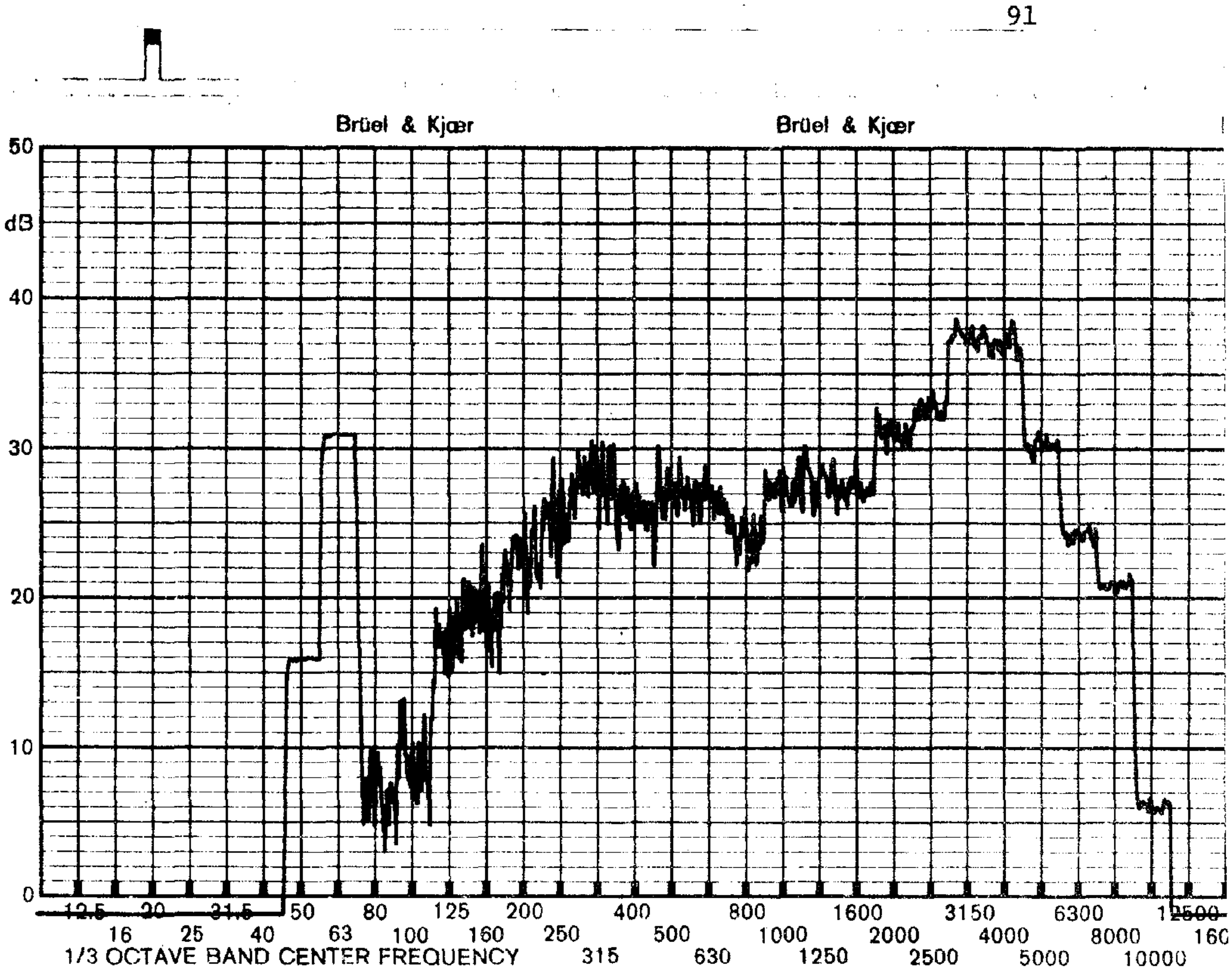

Brüel & Kjær
Copenhagen
Measuring Obj.: U.S.BM D371 With 2k Ω Load Resistor 20 dB Pre-Amp
Rec.No.:
Date:
Sign:
Rect.: RMS
Zero Lev.: 20
L. Lim. Fr.: 20
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq. Scale by:
QP 1151
Brüel & Kjær
Brüel & Kjær
50
dB
40
30
20
10
0
12.5 20 31.5 50 80 125 200 400 800 1600 3150 6300 12500
16 25 40 63 100 160 250 500 1000 2000 4000 8000 160
1/3 OCTAVE BAND CENTER FREQUENCY 315 630 1250 2500 5000 10000

Fig. 40

Brüel & Kjær
Copenhagen

Fig. 41
Measuring Obj.: [redacted]
Lapel
Crystal mic
with 20 dB
Pre-amp

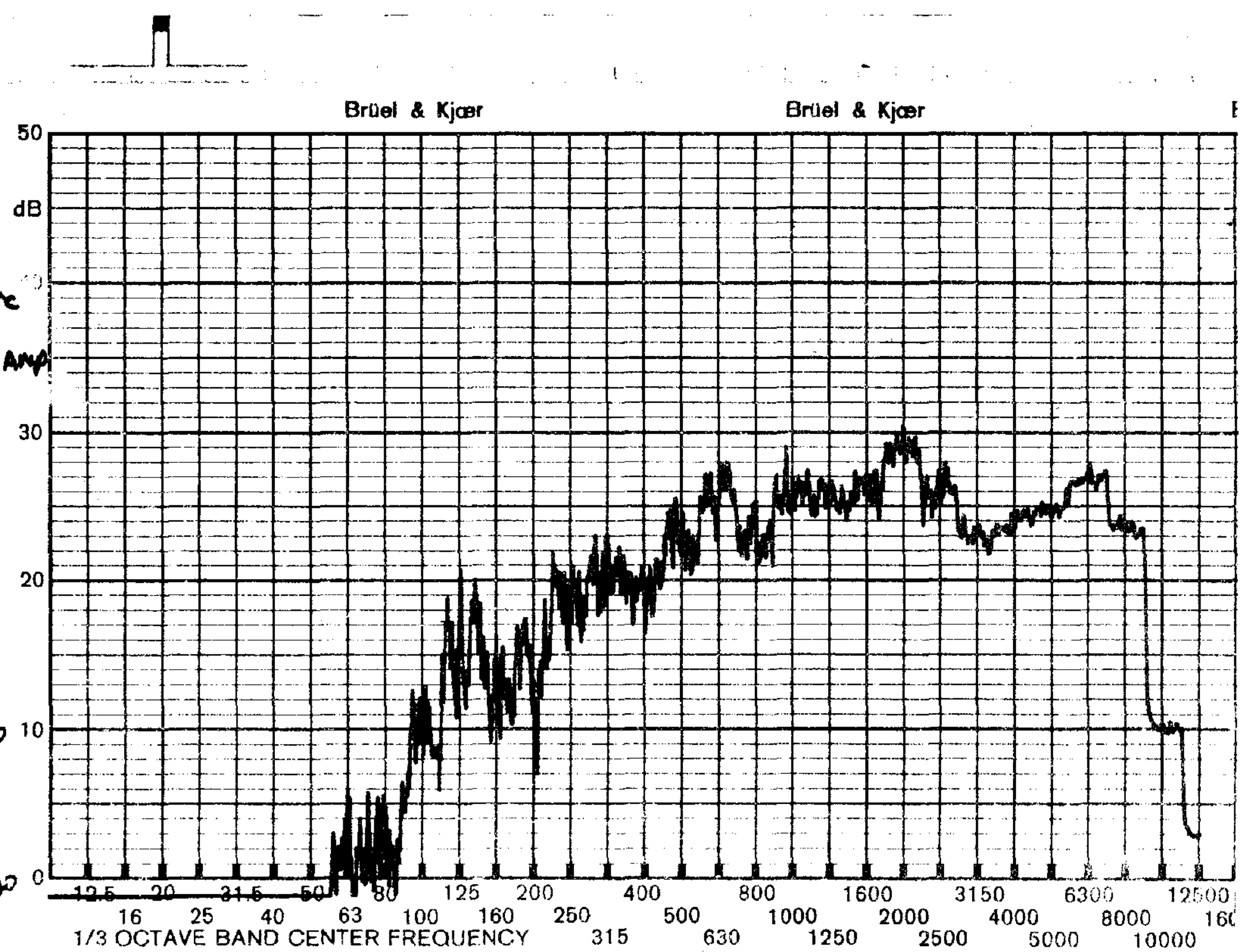
Rec. No.:
Date:
Sign:
Rect.: RMS
Zero Lev.: 50 dB
Lim. Fr.: 20
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq. Scale by:
QP 1151



Brüel & Kjær
Copenhagen

Fig. 42
Measuring Obj.:
$10 Dynamic
Hi Z.
20 dB pre amp

Rec. No.:
Date:
Sign:
Rect.: RMS
Zero Lev.: 20 dB
Lim. Fr.: 20 30
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq. Scale by: 120
QP 1151



Brüel & Kjær Brüel & Kjær Brüel & Kjær

Brüel & Kjær
Copenhagen

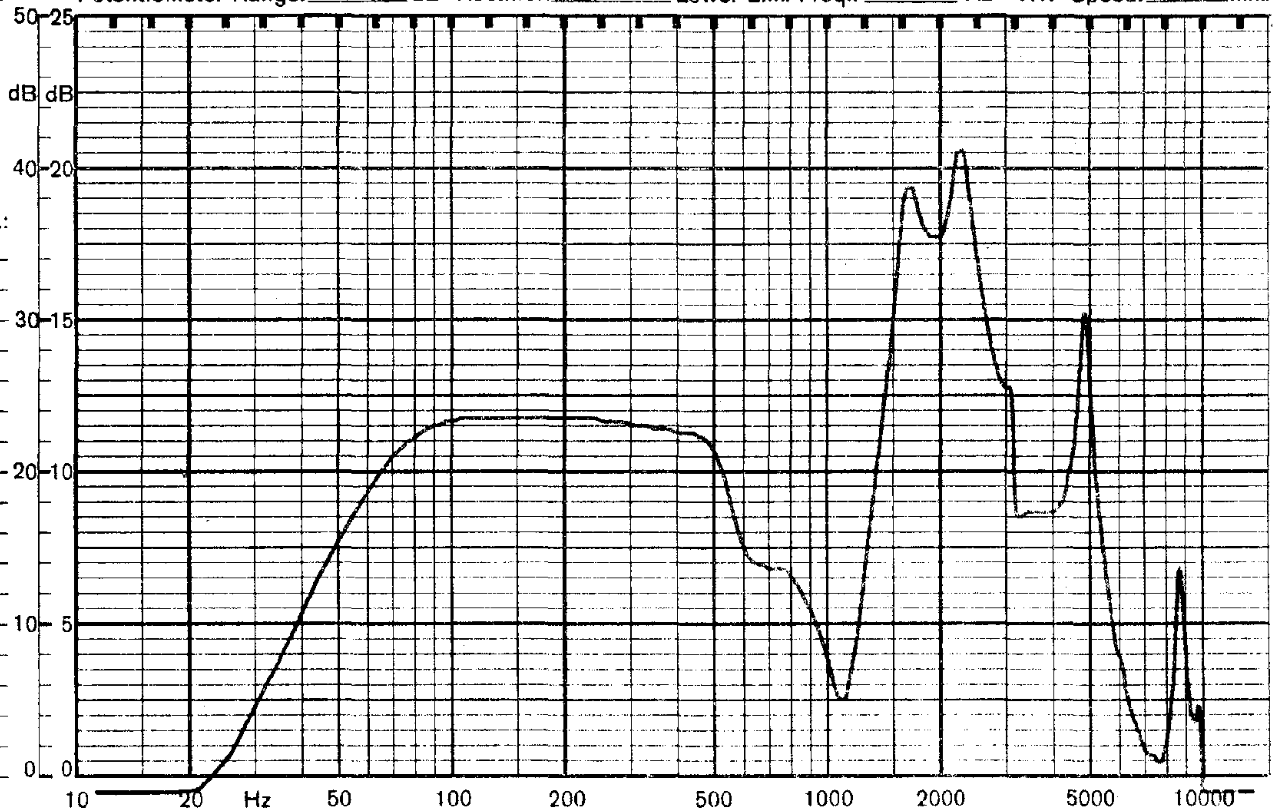
Potentiometer Range: ____ dB Rectifier: ____ Lower Lim. Freq.: ____ Hz Wr. Speed: ____ mm/

Fig. 43

Measuring Obj.: D 426 Earphone in 6cc Coupler

.09 V rms Input

Rec. No.: ____
Date: ____
Sign.: ____



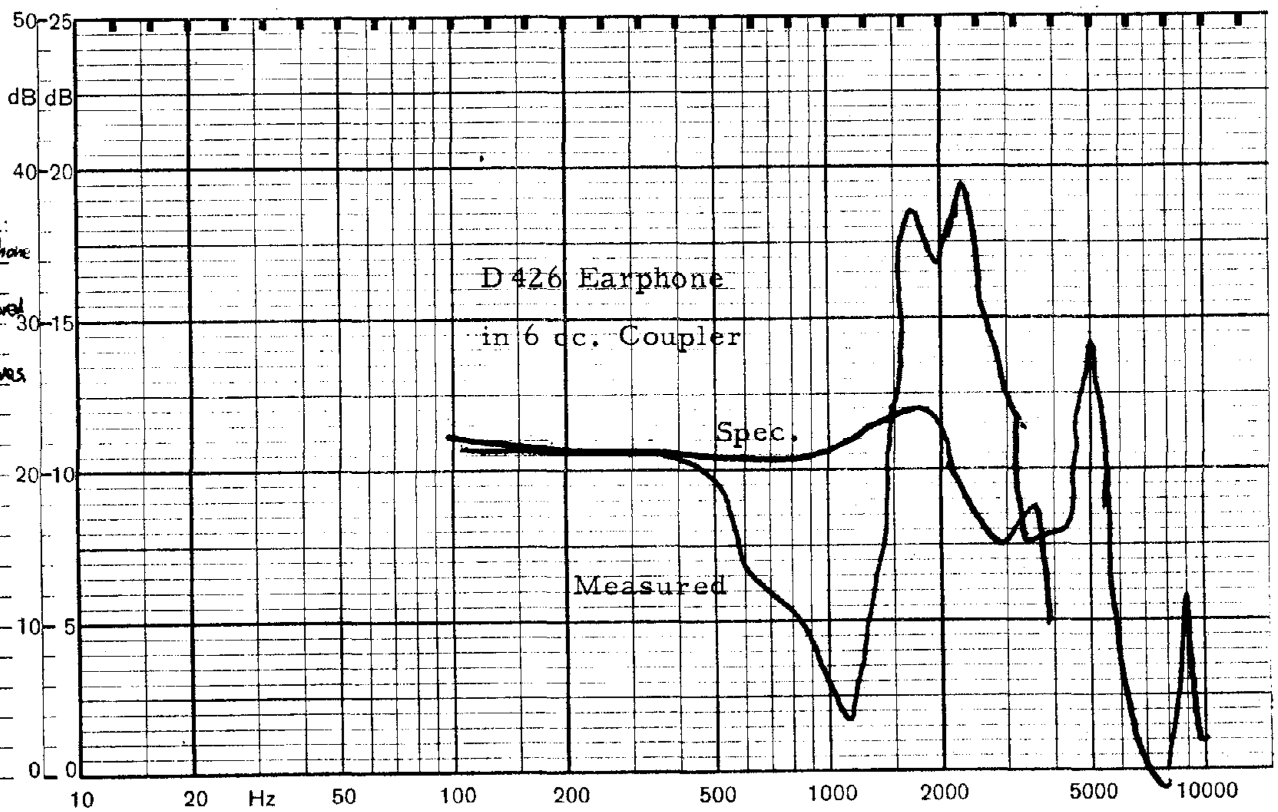
QP 1124 Multiply Frequency Scale by Zero Level: (161

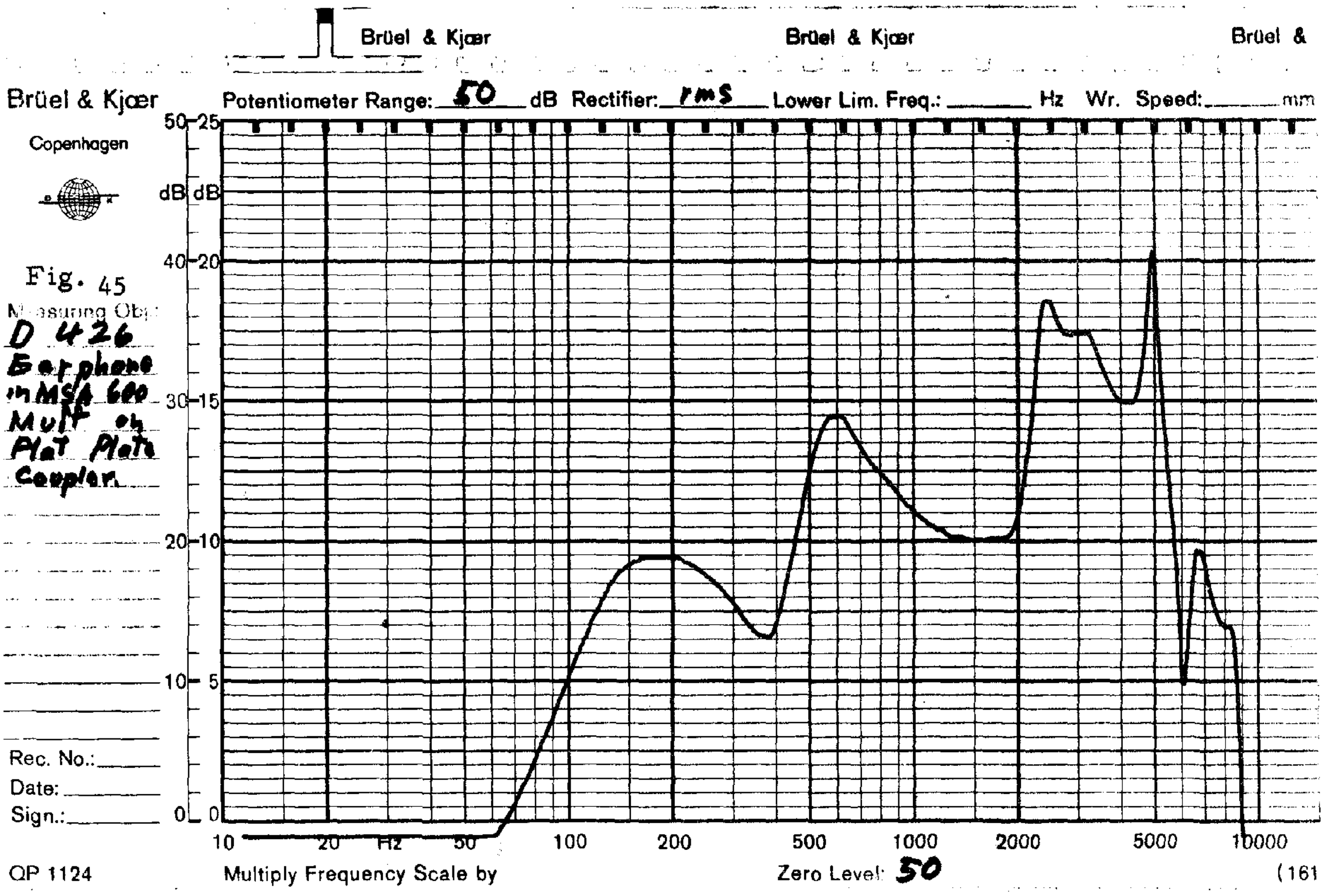
Copenhagen

Fig. 44

Measuring Obj.: D 426 Earphone Comparison between Measured and published Response curves

Rec. No.: ____
Date: ____
Sign.: ____





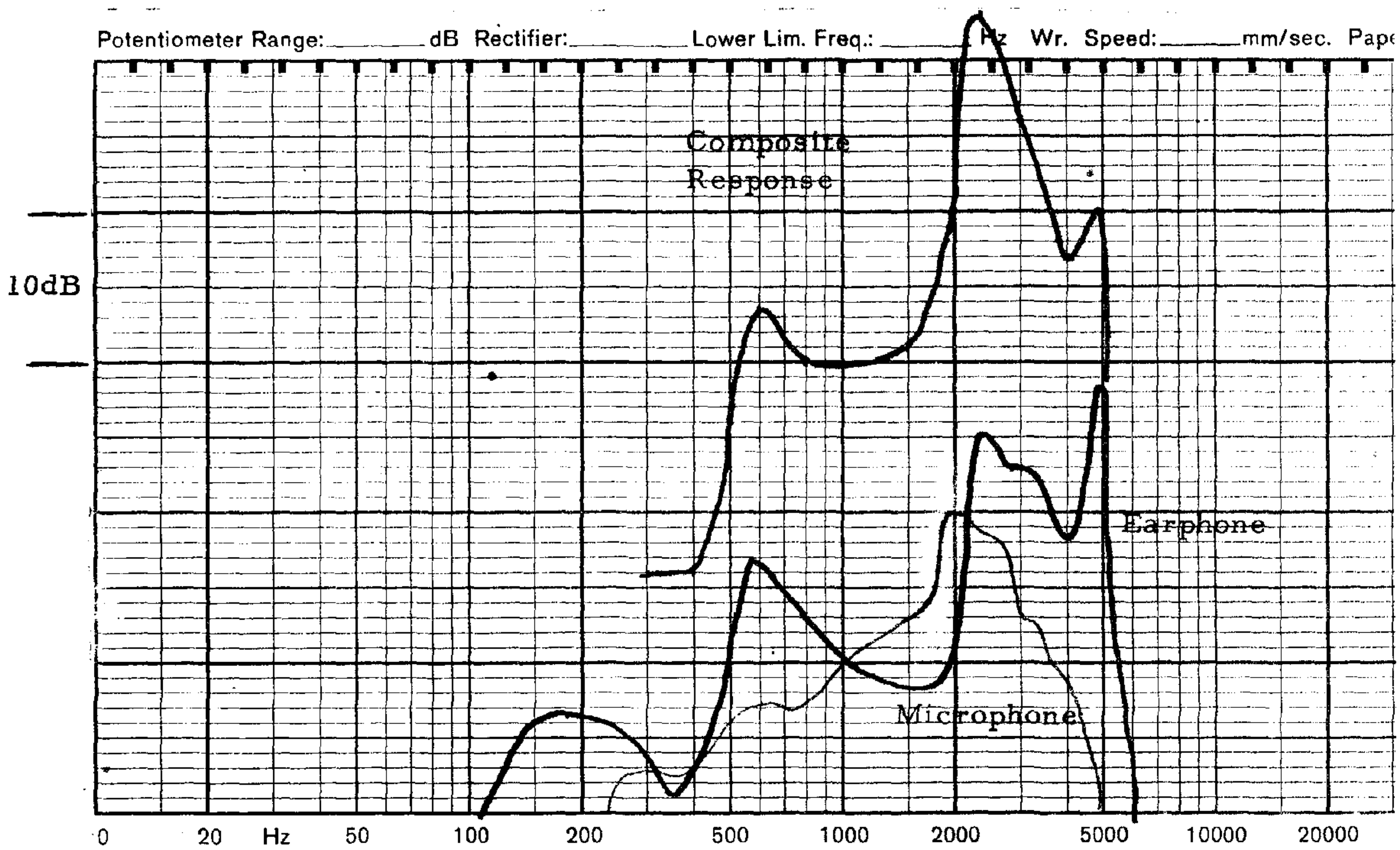


Fig. 46 Composite Response of USBM System from Response of Components

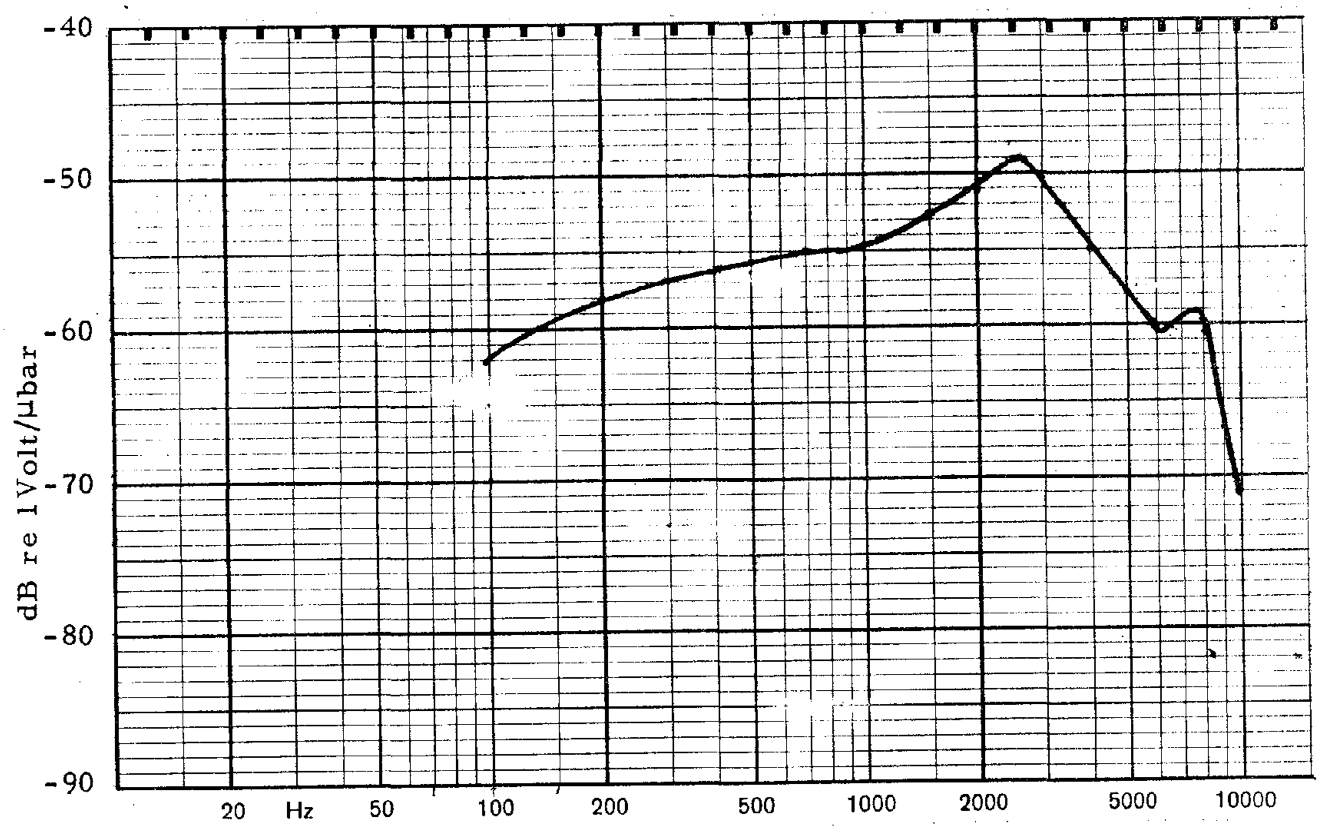


FIG. 47 Manufacturer's specifications for sensitivity of B L-1671 mic.

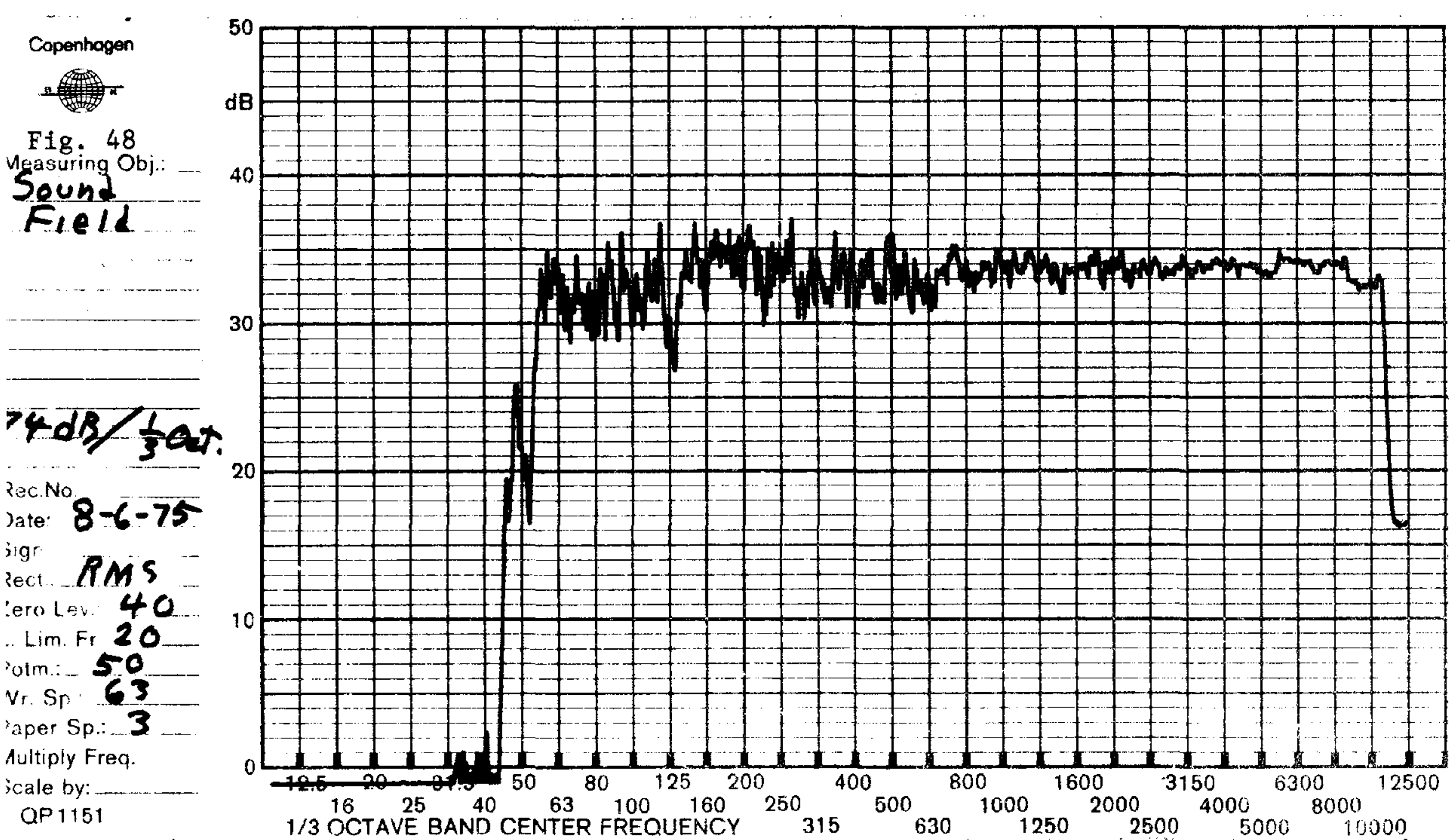


Fig. 48

Fig. 49

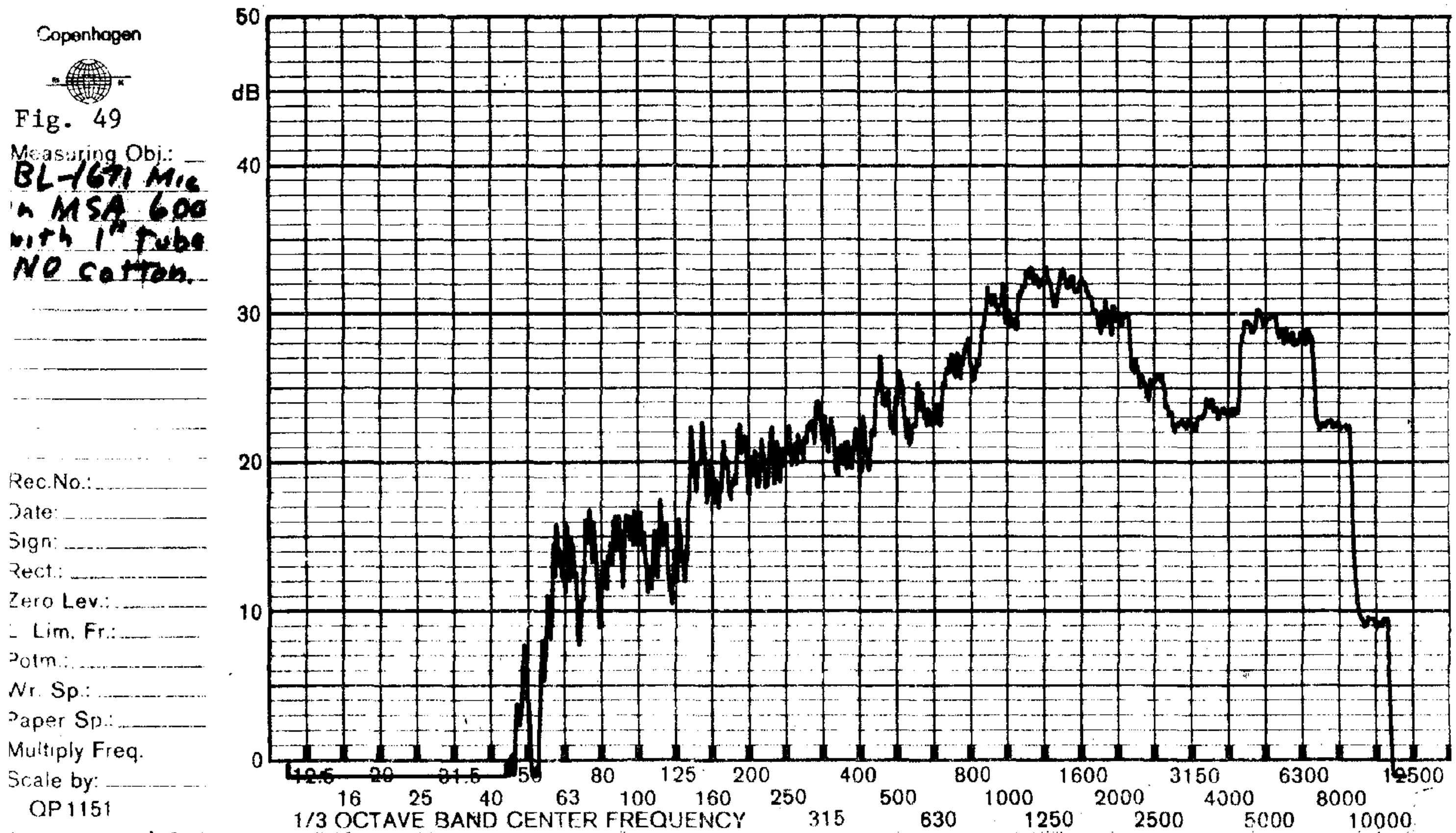
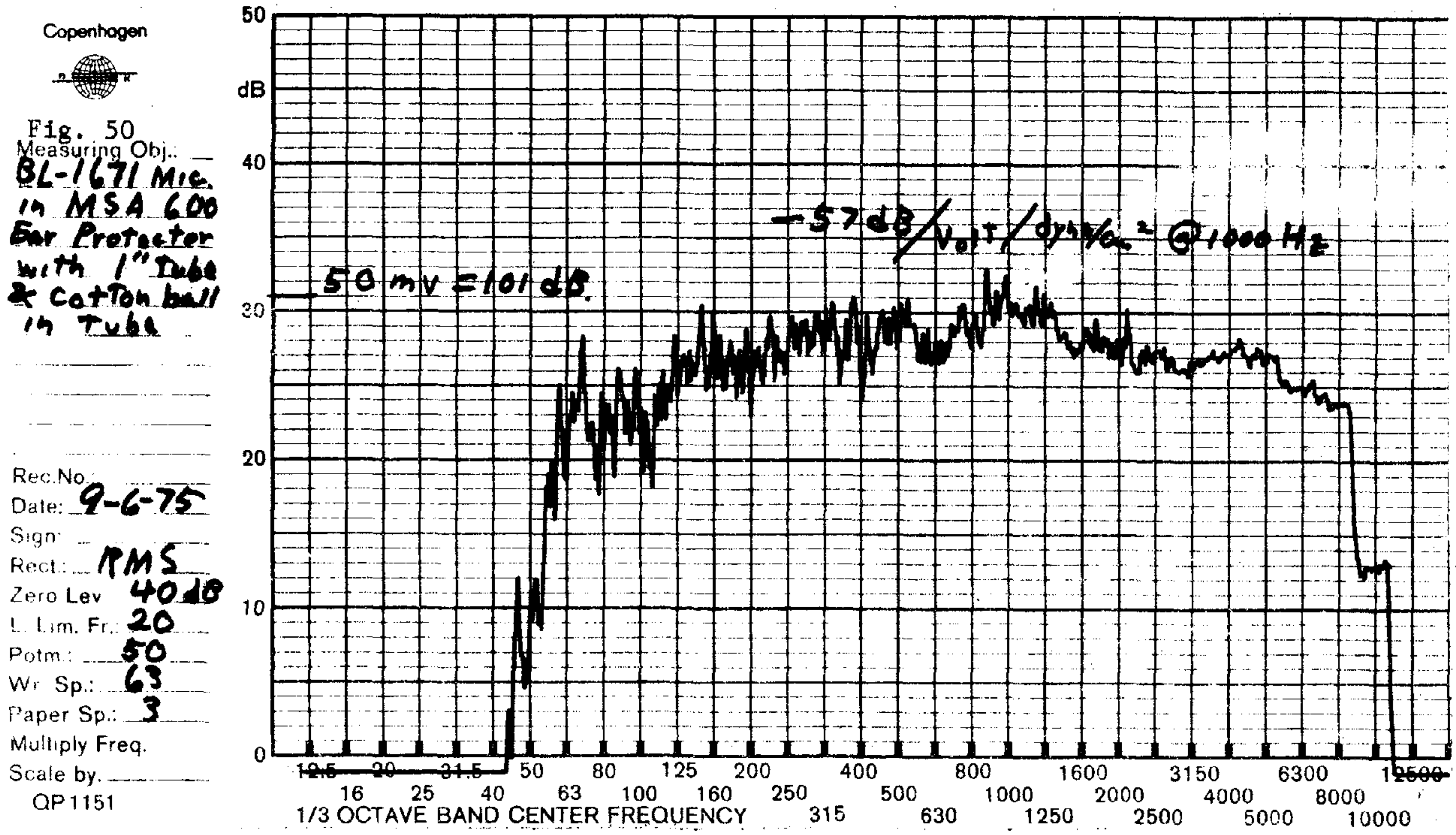

Copenhagen
Measuring Obj.: BL-1671 Mic in MSA 600 with 1" Tube NO cotton.
Rec.No.:
Date:
Sign:
Rect.:
Zero Lev.:
L. Lim. Fr.:
Potm.:
Wr. Sp.:
Paper Sp.:
Multiply Freq.
Scale by:
QP 1151
50
dB
40
30
20
10
0
1/3 OCTAVE BAND CENTER FREQUENCY

Fig. 50


Copenhagen
Measuring Obj.: BL-1671 Mic. in MSA 600 Ear Protecter with 1" Tube & Cotton ball in Tube
Rec.No.
Date: 9-6-75
Sign:
Rect.: RMS
Zero Lev 40 dB
L. Lim. Fr.: 20
Potm.: 50
Wr. Sp.: 63
Paper Sp.: 3
Multiply Freq.
Scale by.
QP 1151
-57 dB/Volt/dyn/cm² @ 1000 Hz
50 mv = 101 dB
50
dB
40
30
20
10
0
1/3 OCTAVE BAND CENTER FREQUENCY

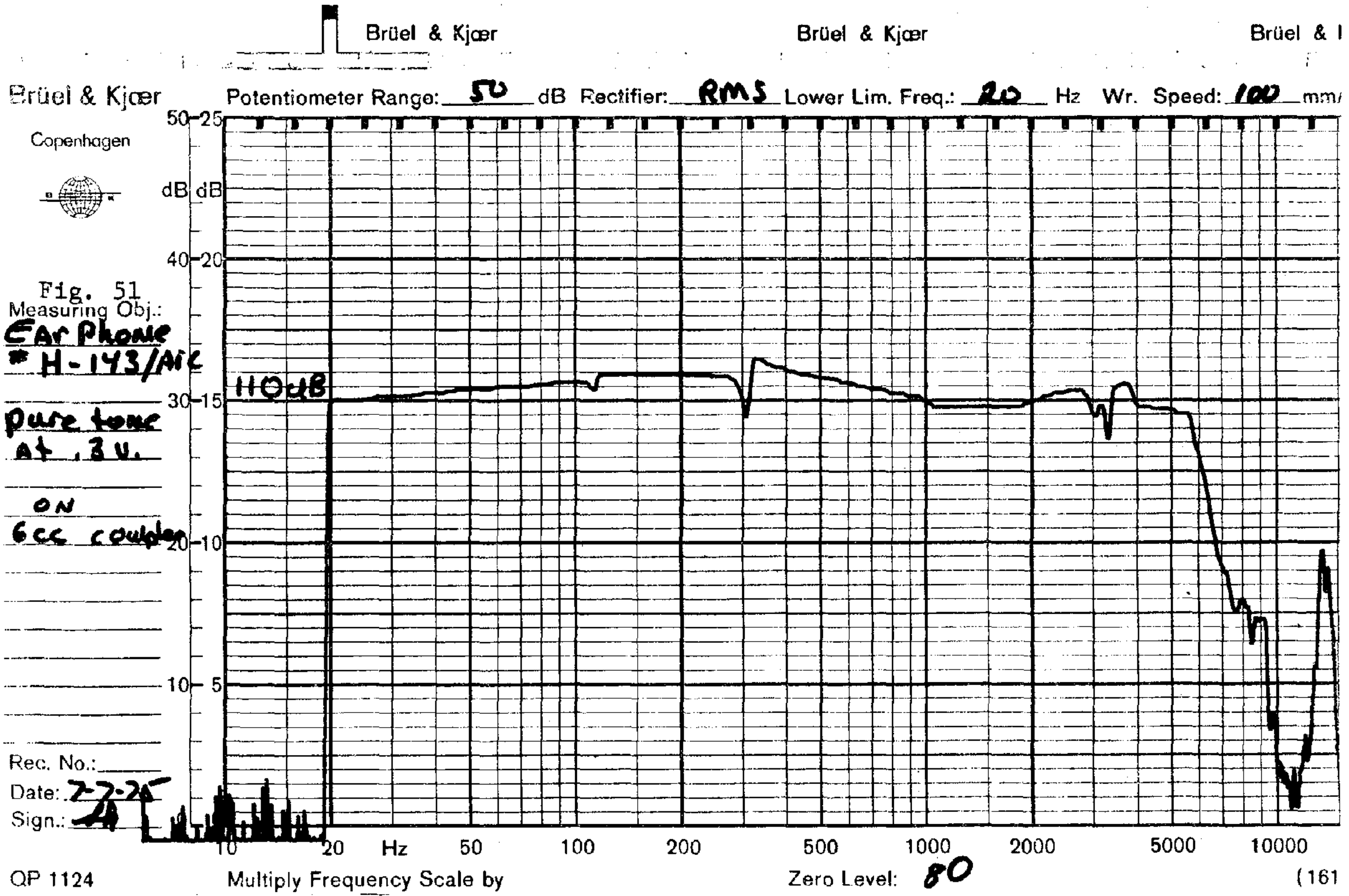

Brüel & Kjær
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm/
Measuring Obj.:
Ear Phone
H-143/AIC
pure tone
at .3 v.
ON
6cc coupler
110dB
Rec. No.:
Date: 7-7-75
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 80

Fig. 51

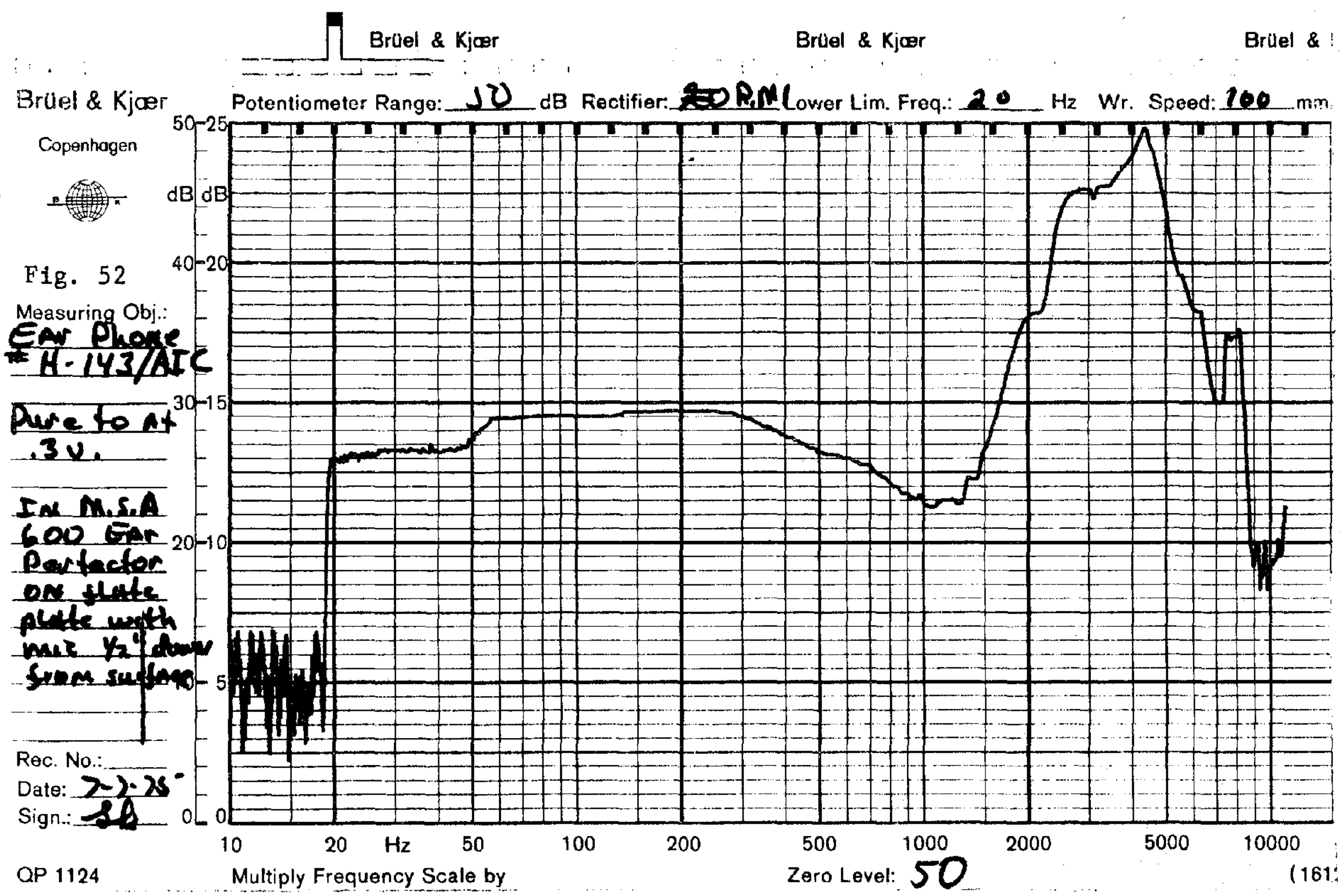

Brüel & Kjær
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm
Measuring Obj.:
Ear Phone
H-143/AIC
Pure to at
.3 v.
In M.S.A
600 Ear
Perfector
ON slate
plate with
mic ½" down
from surface
Rec. No.:
Date: 7-7-75
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 50

Fig. 52

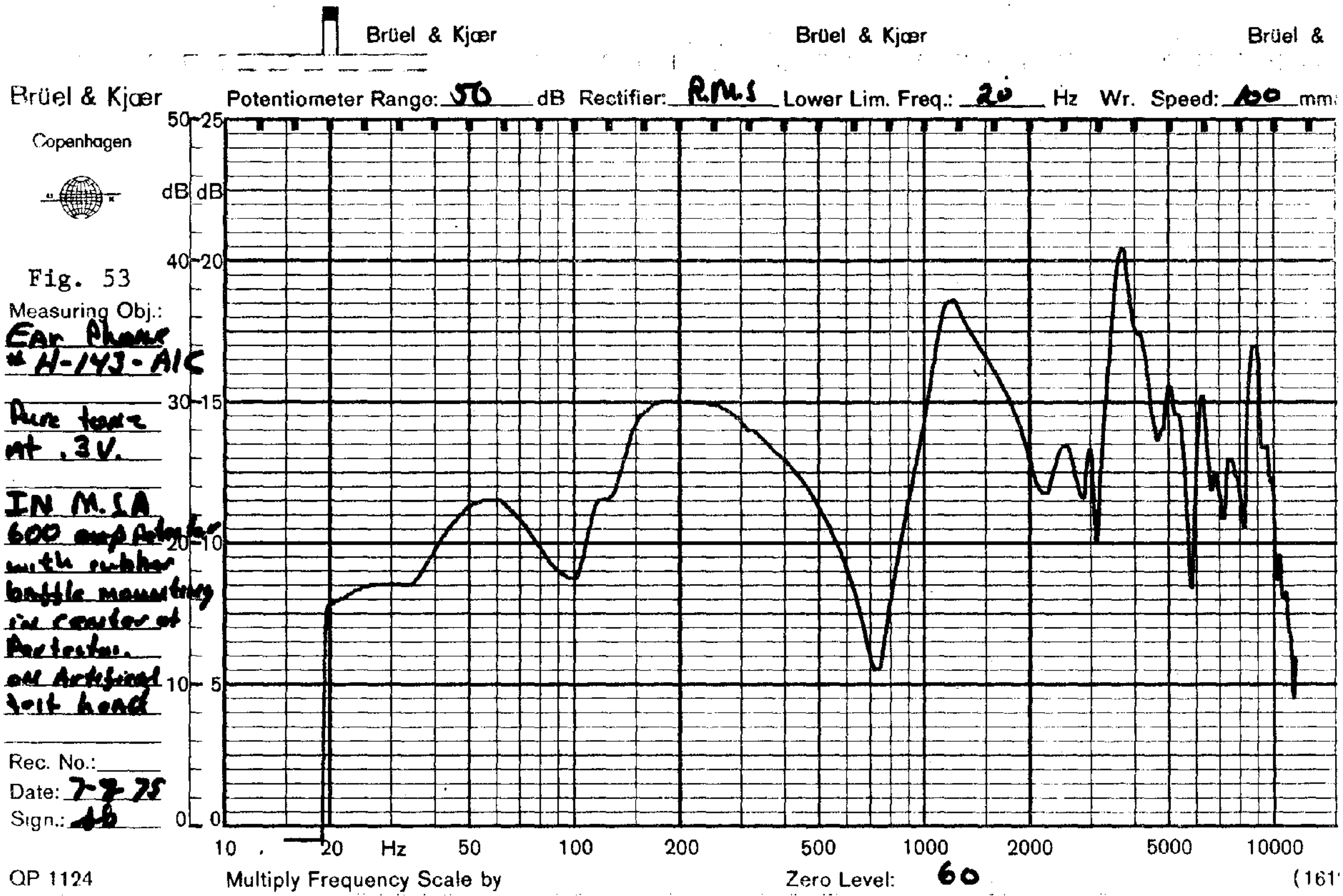

Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm
Measuring Obj.:
Ear Phone #H-143-A1C
Pure tone at .3V.
IN M.S.A 600 ear protector with rubber baffle mounting in center of Protector. on Artificial test head
Rec. No.:
Date: 7-7-75
Sign.:
dB dB
50 25
40 20
30 15
20 10
10 5
0 0
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 60
(161

Fig. 53

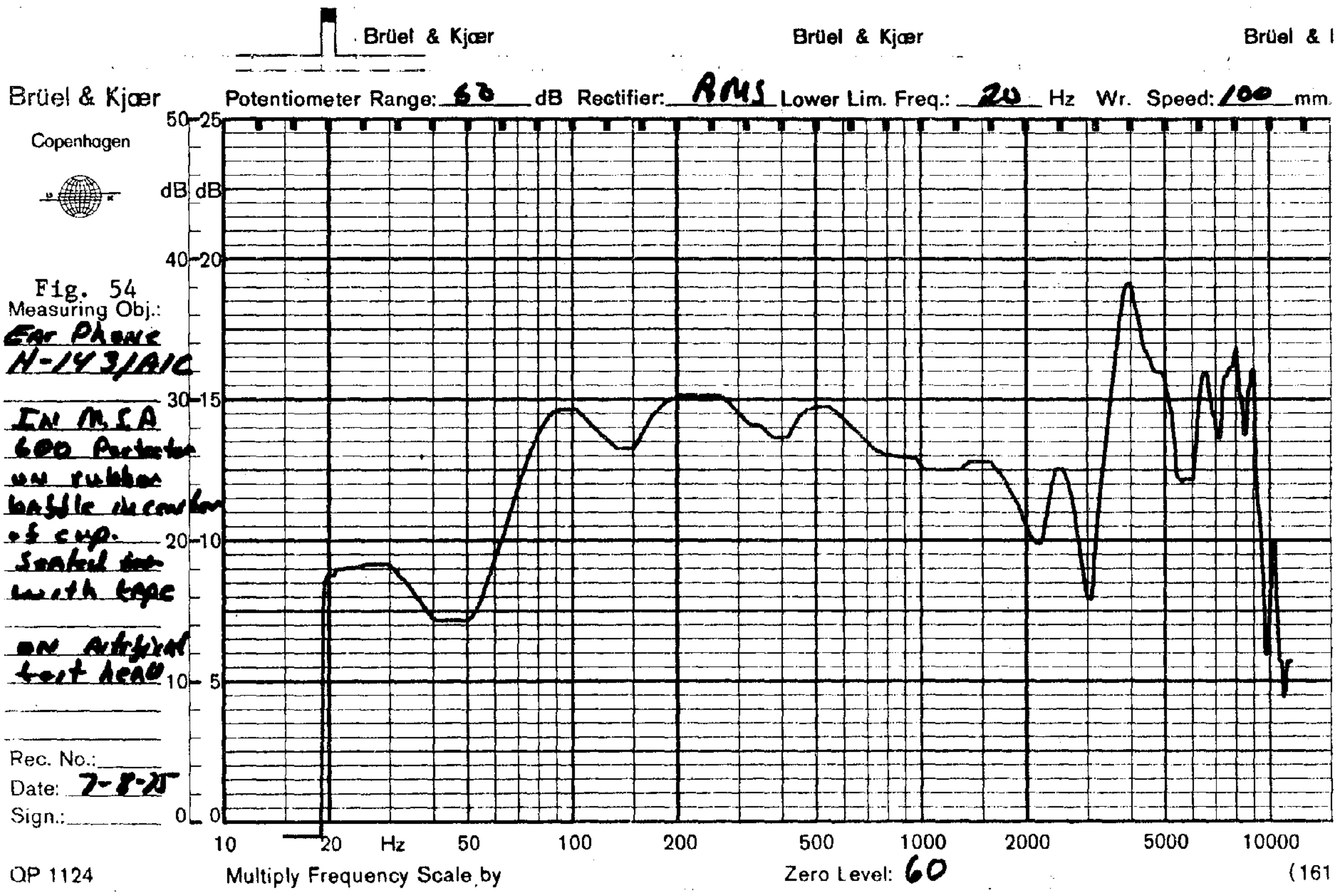

Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm
Measuring Obj.:
Ear Phone H-143/A1C
IN M.S.A 600 Protector on rubber baffle in center of cup. Sealed with tape
on Artificial test head
Rec. No.:
Date: 7-8-75
Sign.:
dB dB
50 25
40 20
30 15
20 10
10 5
0 0
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 60
(161

Fig. 54

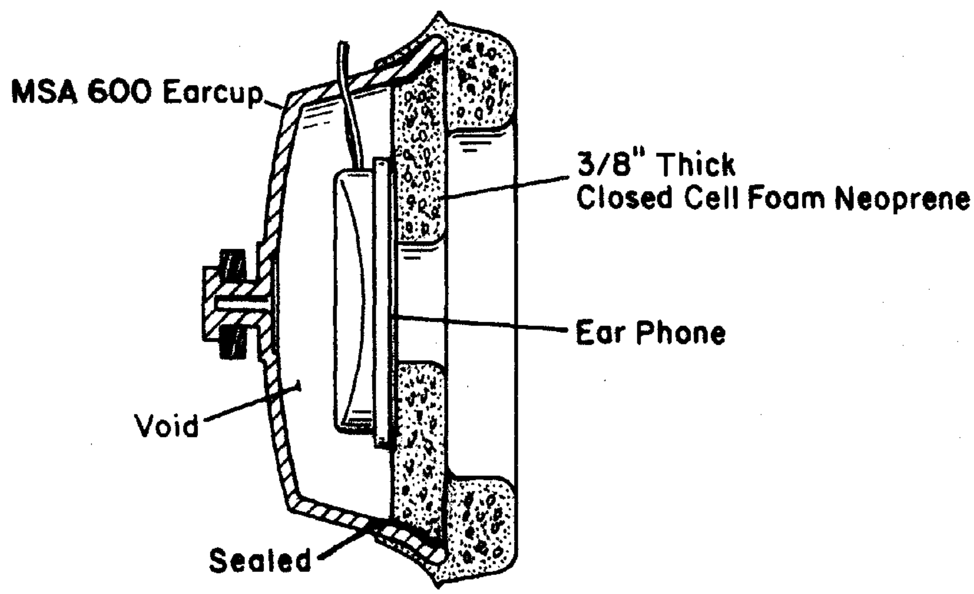


Fig. 55 Earphone Mounted on 3/8" closed cell Neoprene.

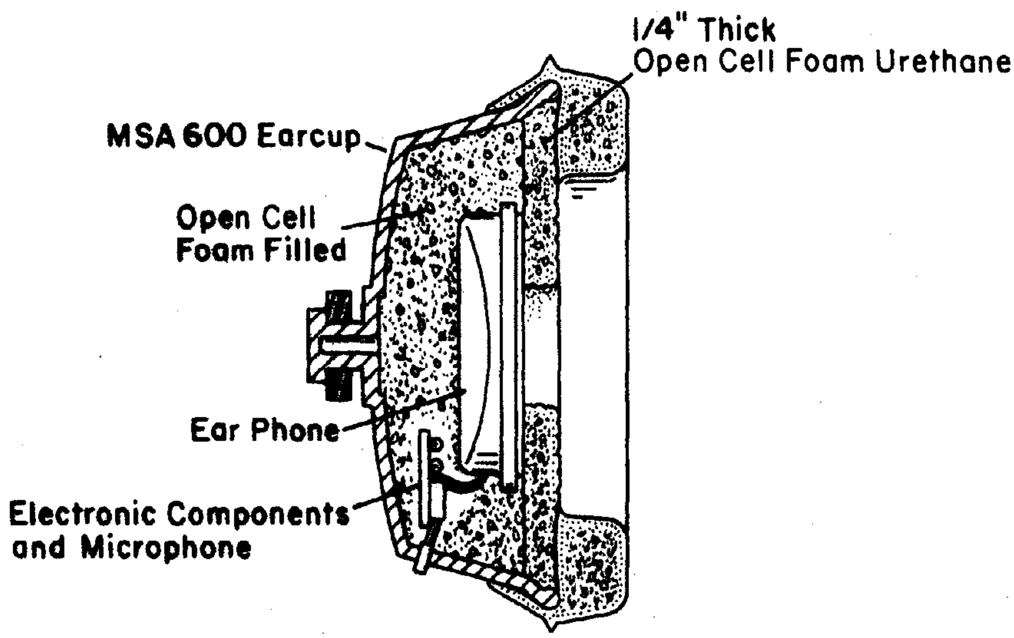


Fig. 56 Earphone and components mounted in open cell foam.

Brüel & Kjær Copenhagen

Potentiometer Range: 50 dB Rectifier: R.M.S Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm/

Fig. 57

Measuring Obj.: Ear Phone # H-143 A/C

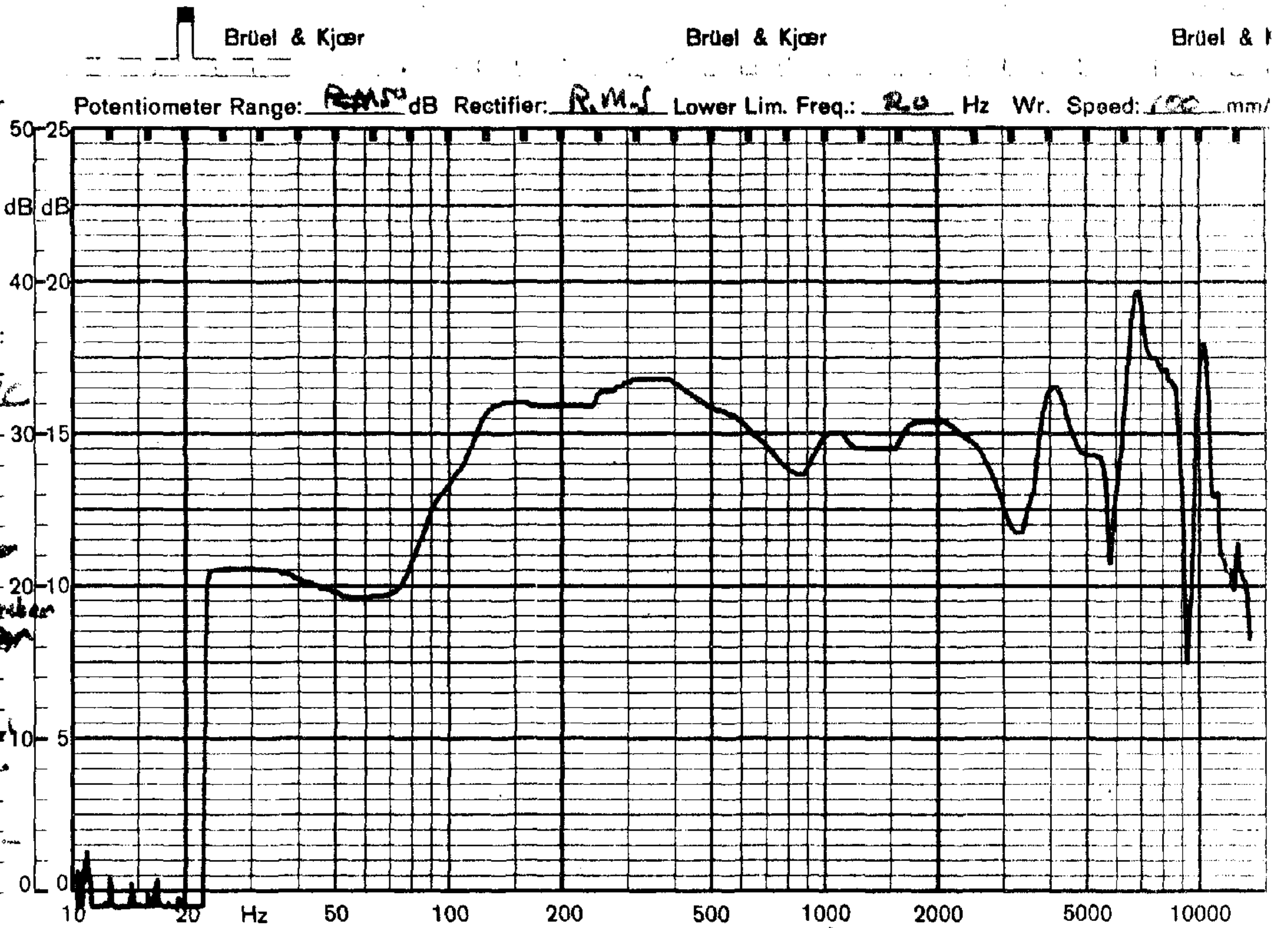
Mounted on baffle in M.S.A 600. Rubber sealed to make chamber behind ear phone.

On artificial test head. input .3 V.

Rec. No.:

Date: 7-11-75

Sign.: db



QP 1124 Multiply Frequency Scale by Zero Level: 60 (161

Brüel & Kjær Copenhagen

Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm

Fig. 58

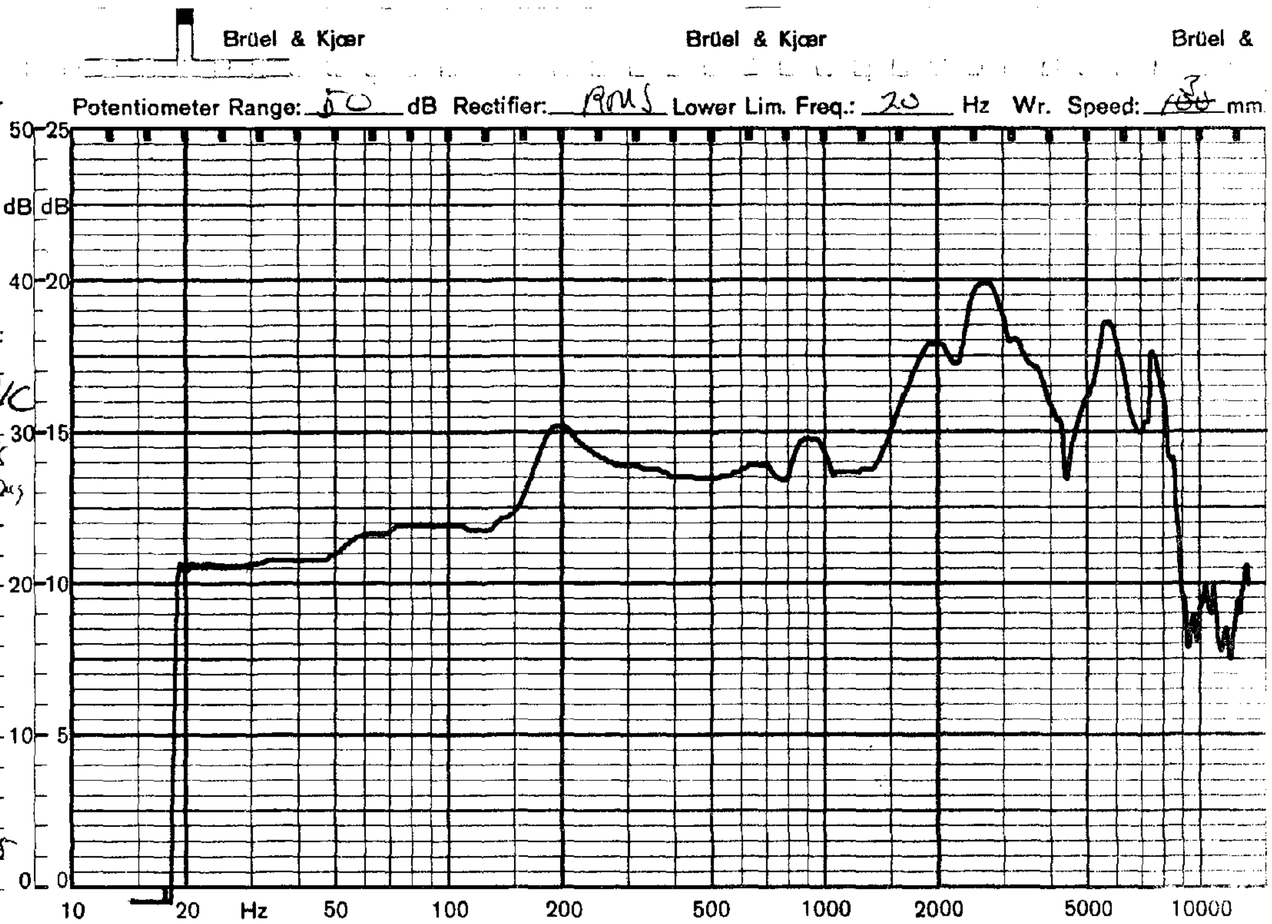
Measuring Obj.: Ear Phone # H-143 /A/C Mounted in open cell foam, Sallys Earlie Ear Detector

.3 V. input on Artifical test head.

Rec. No.:

Date: 7-21-75

Sign.: db



QP 1124 Multiply Frequency Scale by Zero Level: 60 (161

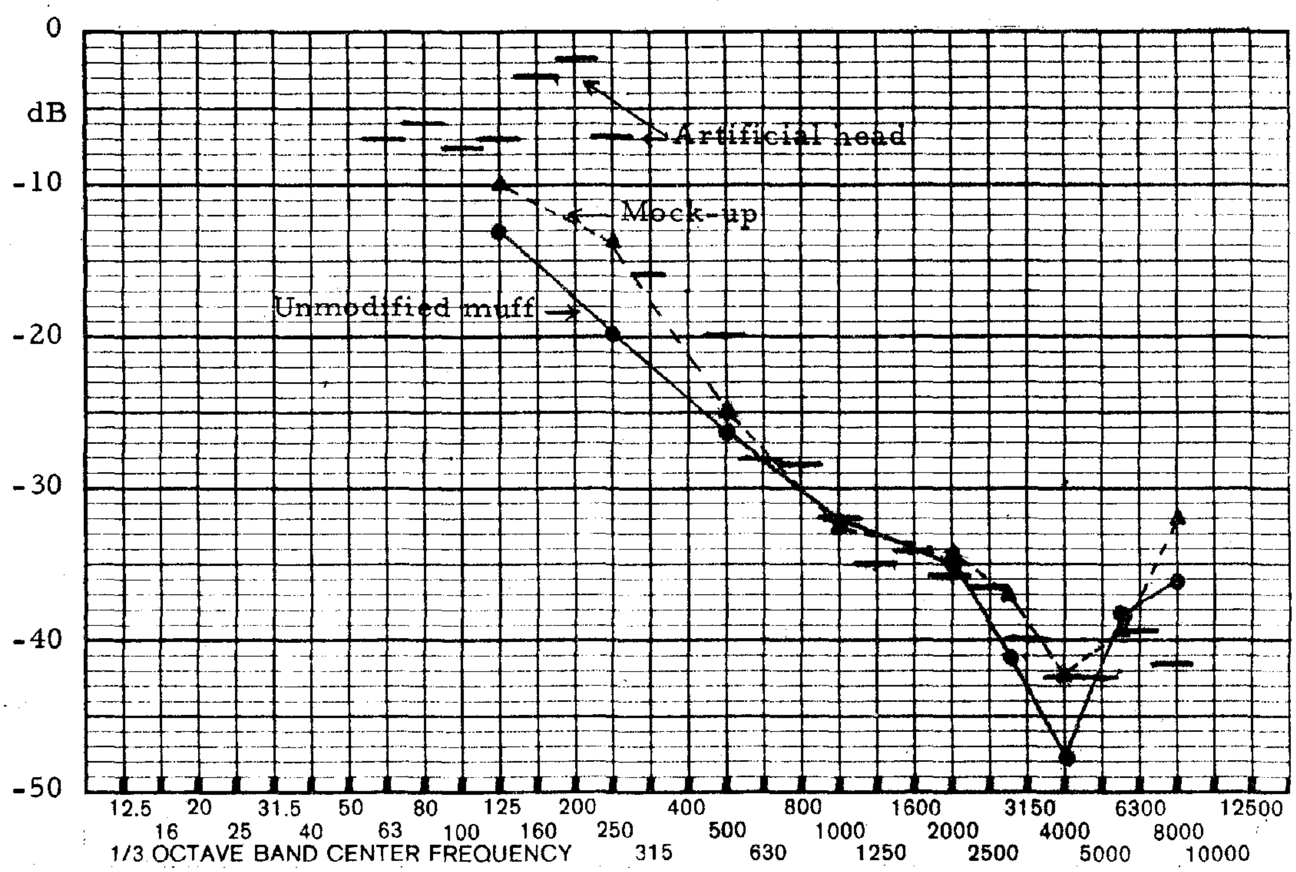
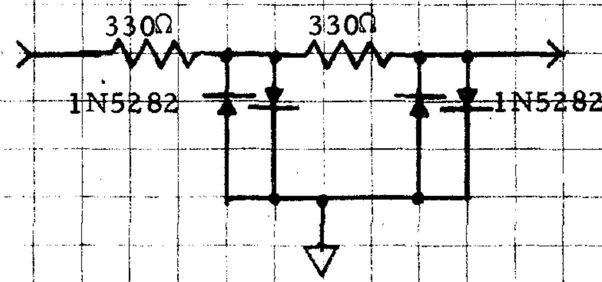
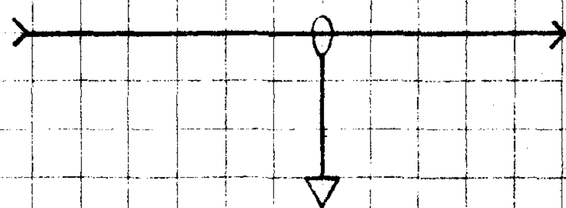


Fig. 59 Real-ear attenuation of hearing protector mock-up based on MSA Comfo 600 hearing protector.

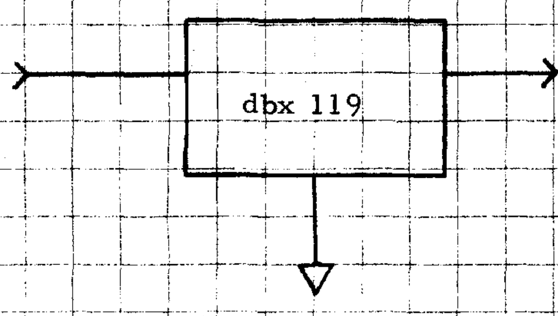
FIGURE 60



NETWORK 1:
TWO STAGE DIODE CLIPPER



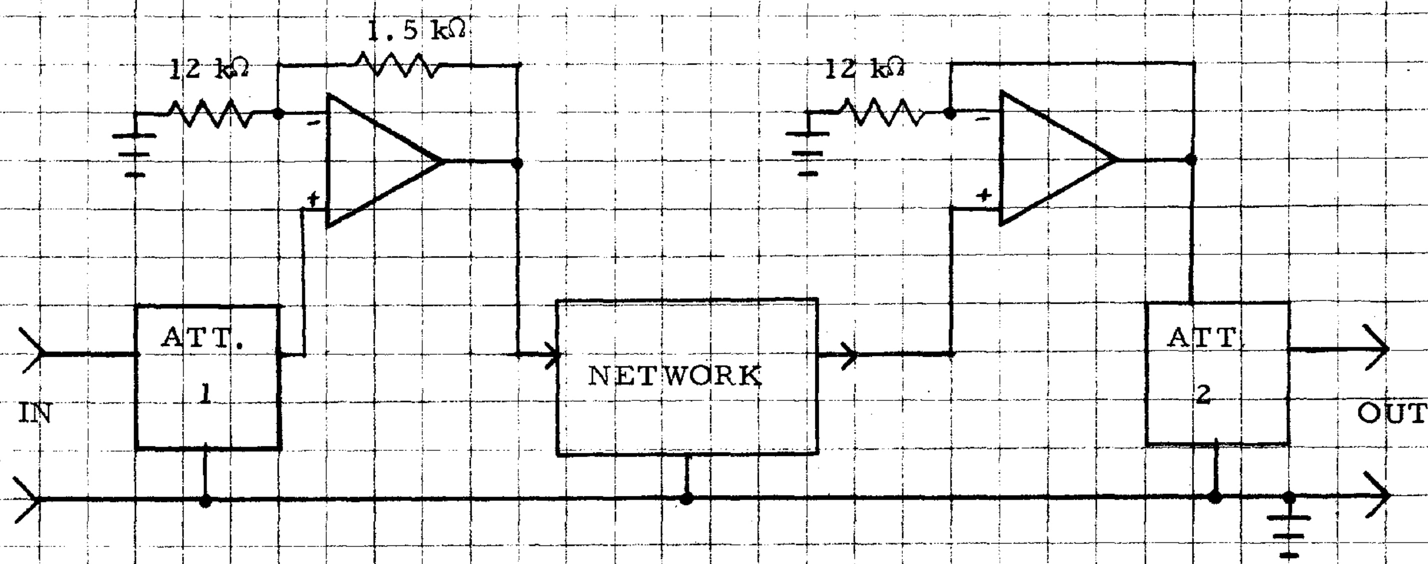
NETWORK 2:
THRU CONNECTION (CONTROL)



NETWORK 3:
SIGNAL LEVEL COMPRESSOR

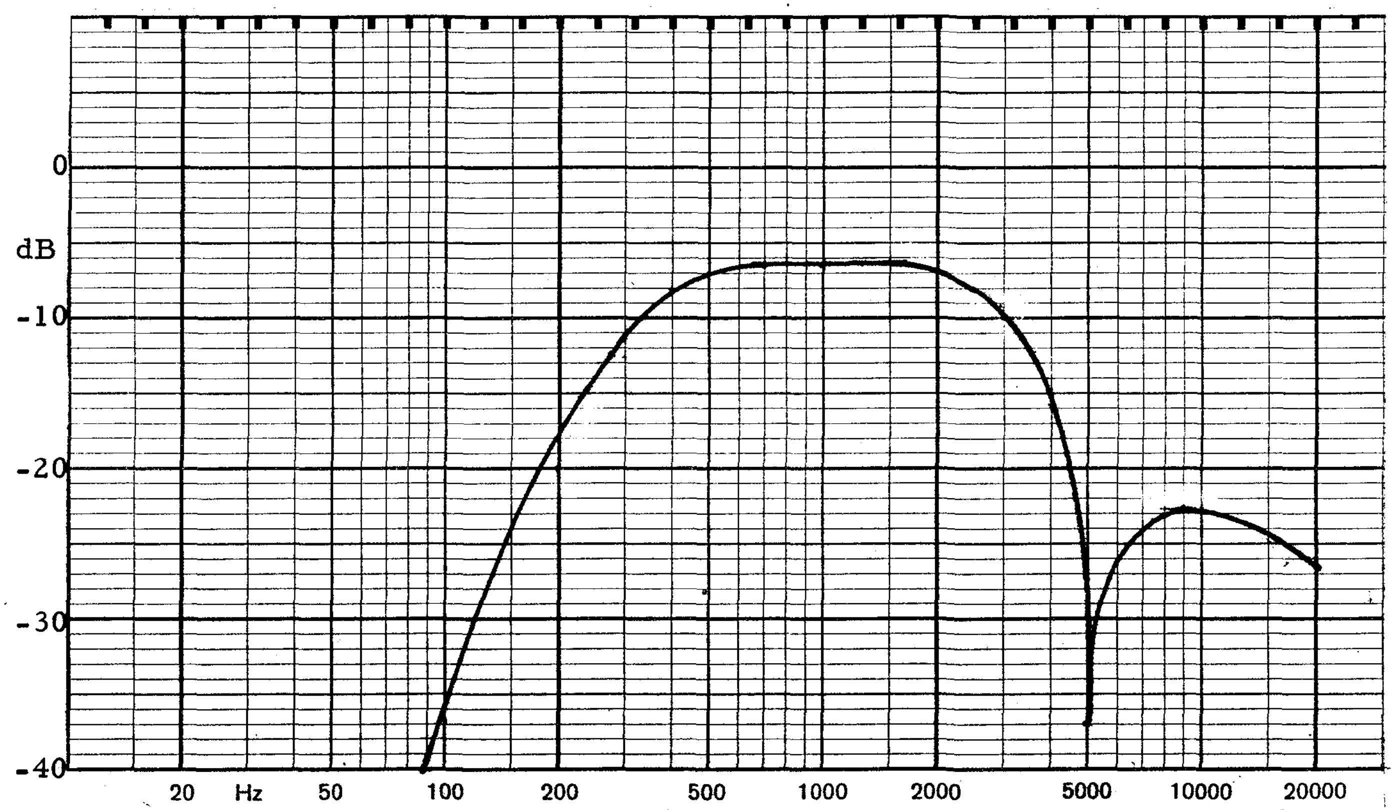
SIGNAL COMPRESSION NETWORKS

FIGURE 61



TEST CIRCUIT FOR COMPRESSION NETWORKS

FIGURE 62



RESPONSE OF SPEECH FILTER

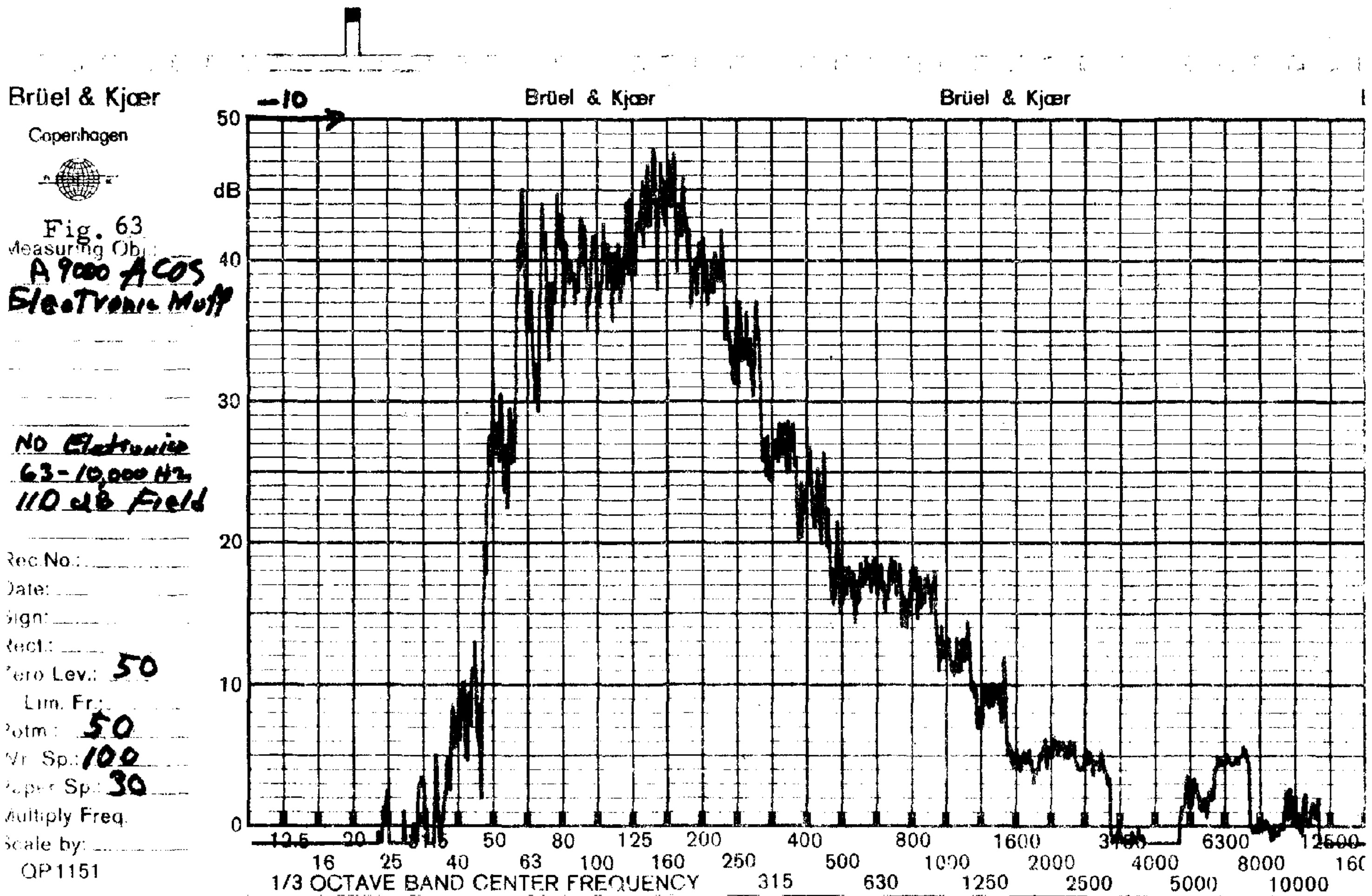

Brüel & Kjær
Copenhagen
Measuring Obj.:
A 9000 ACOS
Electronic Muff
No Electronics
63-10,000 Hz
110 dB Field
Rec. No.:
Date:
Sign:
Rect.:
Zero Lev.: 50
Lim. Fr.:
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
QP 1151
-10
Brüel & Kjær
Brüel & Kjær
50
dB
40
30
20
10
0
12.5 20 31.5 50 80 125 200 400 800 1600 3150 6300 12500
16 25 40 63 100 160 250 500 1000 2000 4000 8000 16000
1/3 OCTAVE BAND CENTER FREQUENCY 315 630 1250 2500 5000 10000

Fig. 63

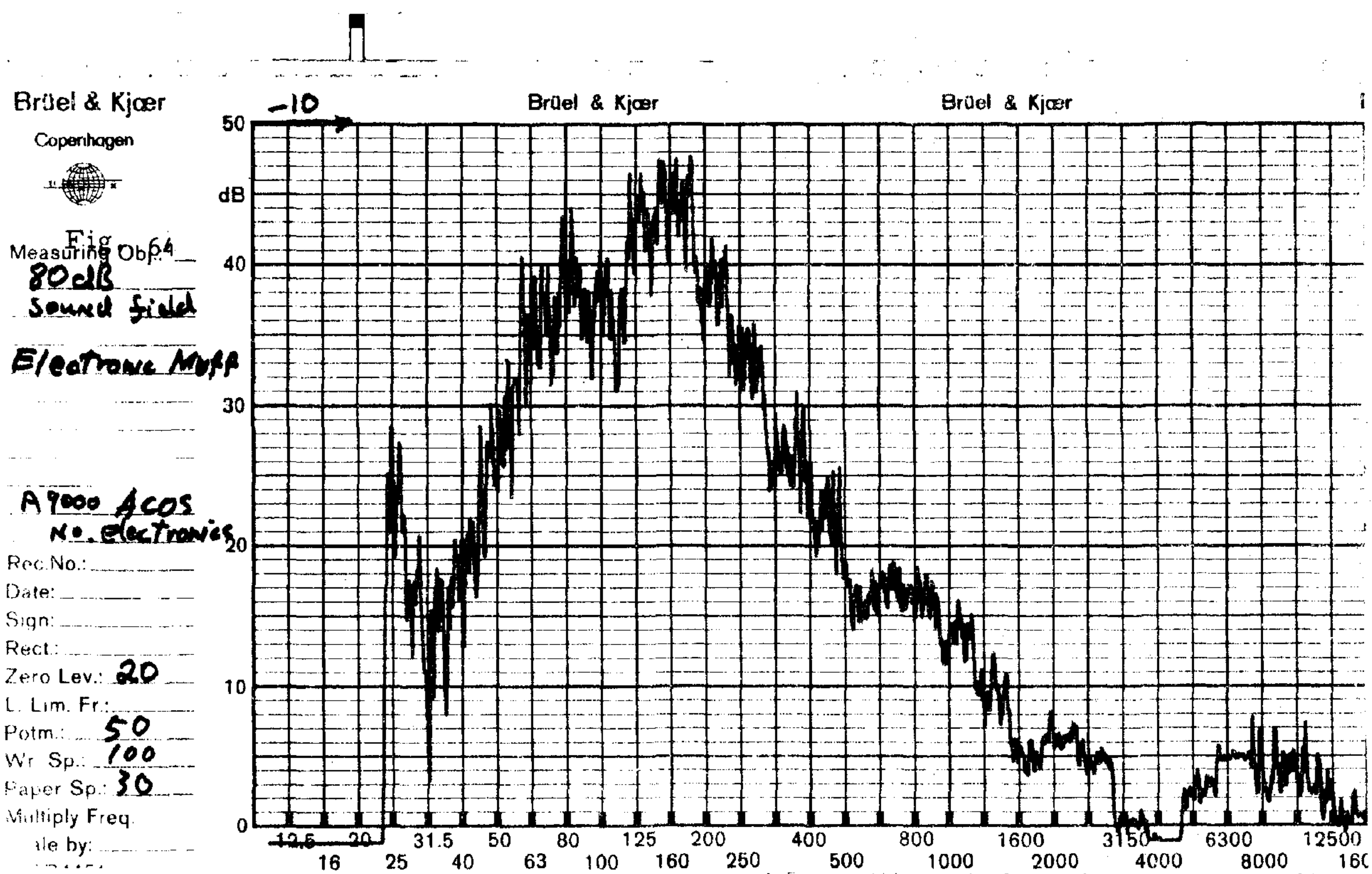

Brüel & Kjær
Copenhagen
Measuring Obj.:
80 dB
Sound field
Electronic Muff
A 9000 ACOS
No electronics
Rec. No.:
Date:
Sign:
Rect.:
Zero Lev.: 20
L. Lim. Fr.:
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
-10
Brüel & Kjær
Brüel & Kjær
50
dB
40
30
20
10
0
12.5 20 31.5 50 80 125 200 400 800 1600 3150 6300 12500
16 25 40 63 100 160 250 500 1000 2000 4000 8000 16000

Fig. 64

Brüel & Kjær
Copenhagen

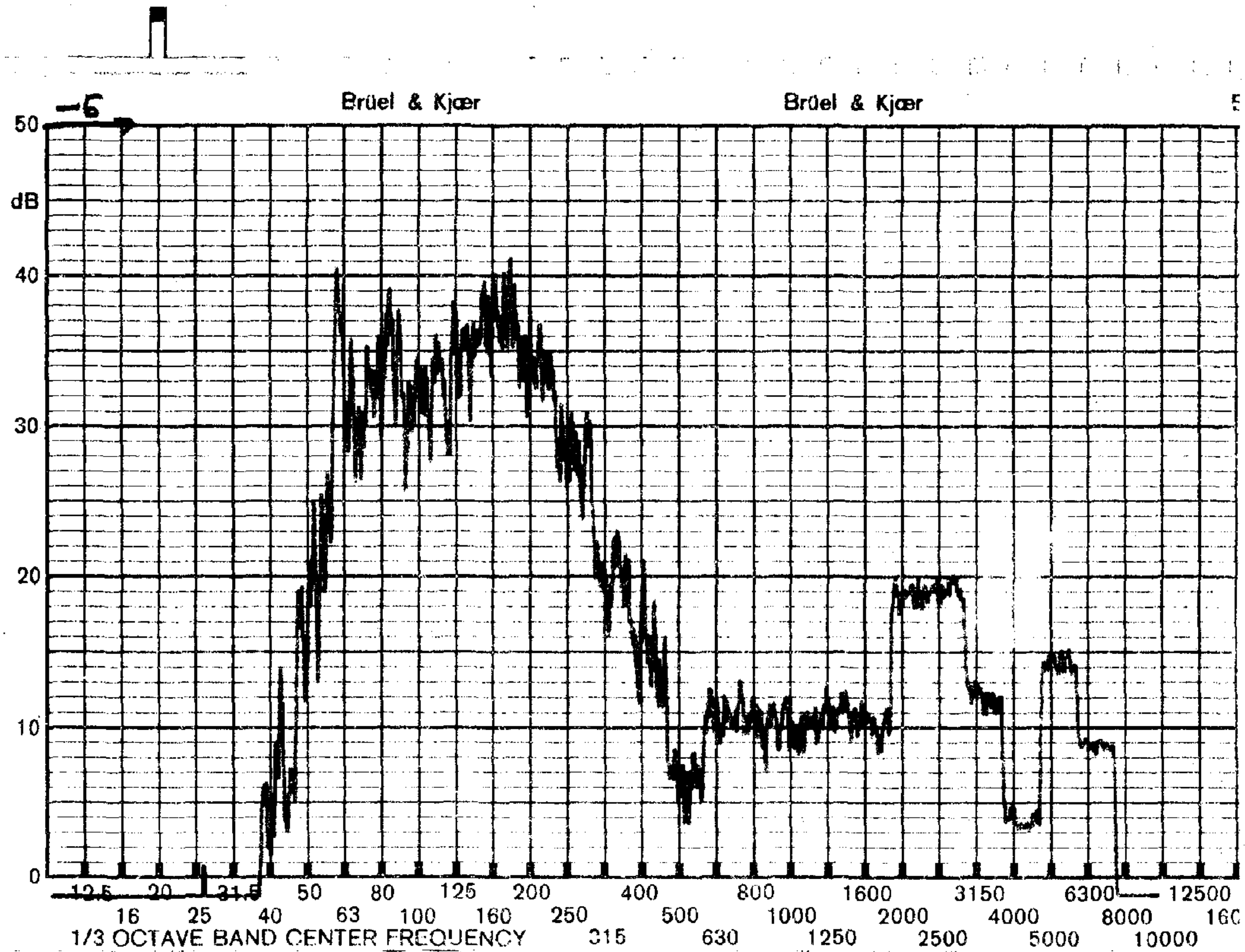
Fig. 65

Measuring Obj.: Sound Field 115 dB

63-10,000 Hz

Glc. on
ACOS
A 9000
Glc. on

Rec. No.:
Date: 5-2-75
Sign: [illegible]
Rect.:
Zero Lev.: 60
L. Lim. Fr.:
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
QP 1151



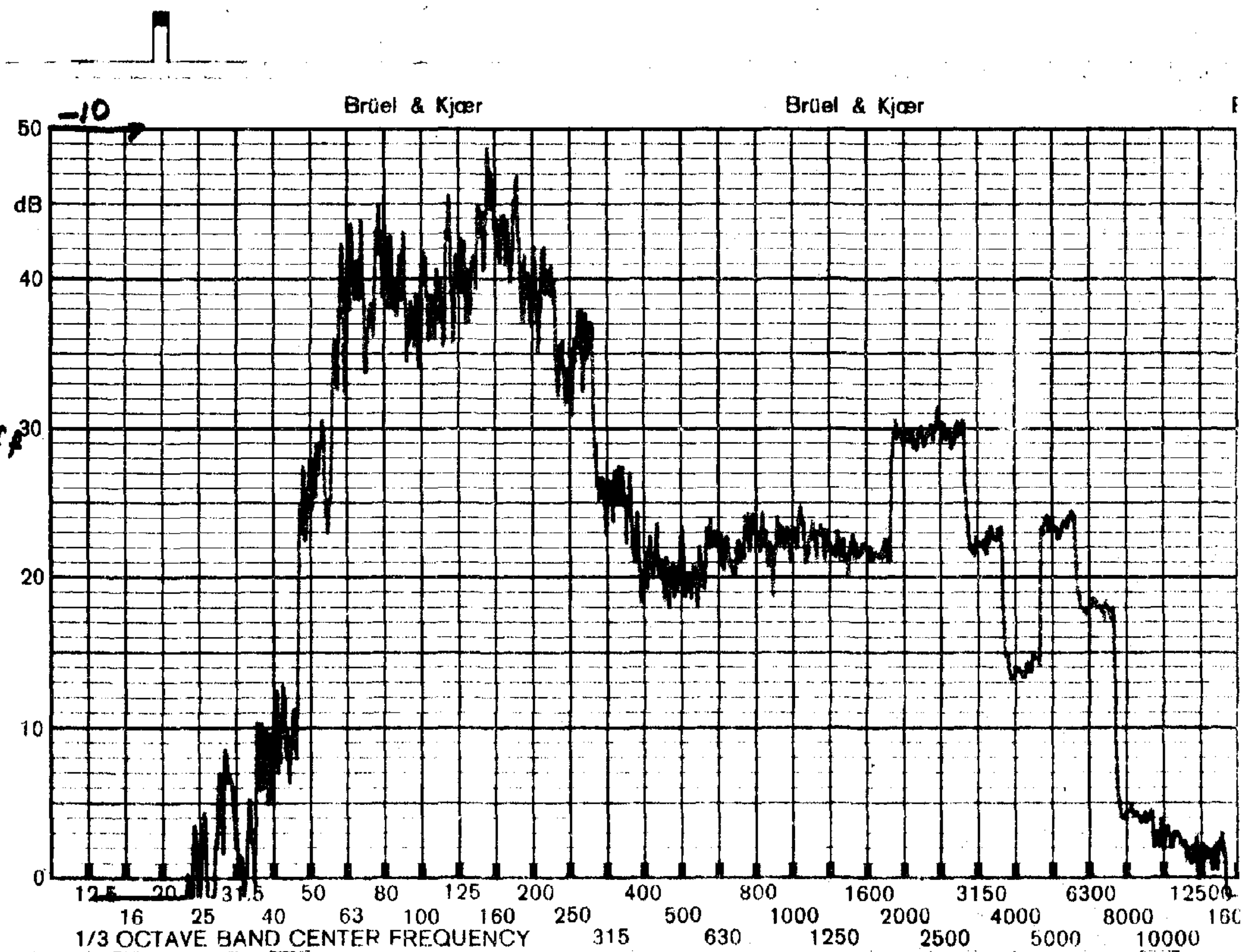
Brüel & Kjær
Copenhagen

Fig. 66

Measuring Obj.: Sound Field 110 dB

Glc. on
ACOS A9000
Electronic Muff

Rec. No.:
Date:
Sign:
Rect.:
Zero Lev.: 50
L. Lim. Fr.:
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
QP 1151



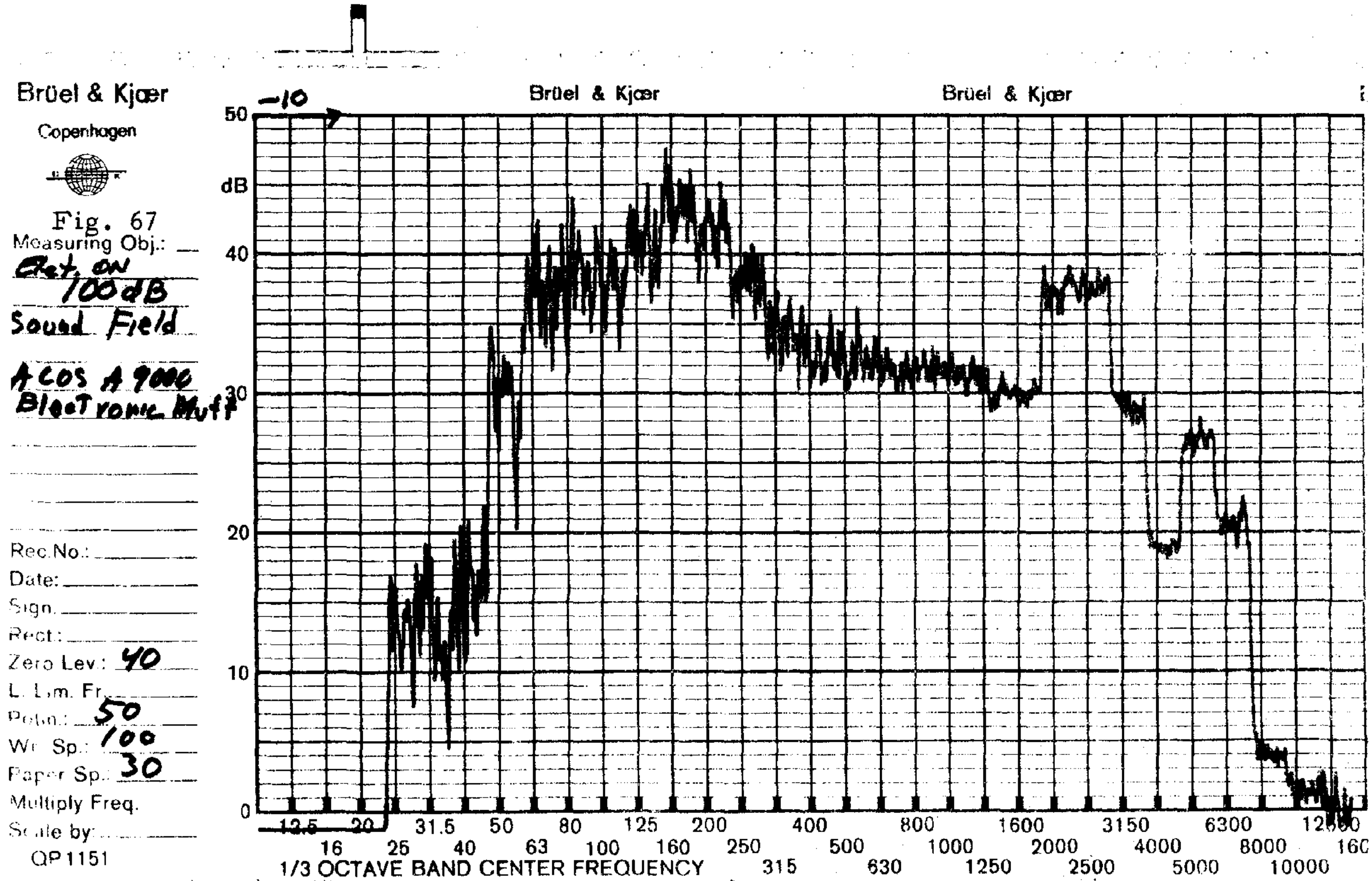

Brüel & Kjær
Copenhagen
Measuring Obj.:
Ext. ON
100 dB
Sound Field
ACOS A 9000
Electronic Muff
Rec.No.:
Date:
Sign.
Rect.:
Zero Lev.: 40
L. Lim. Fr.
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
QP 1151
-10
50
dB
40
30
20
10
0
12.5 16 20 25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 800 1000 1250 1600 2000 2500 3150 4000 5000 6300 8000 10000 12500
1/3 OCTAVE BAND CENTER FREQUENCY

Fig. 67

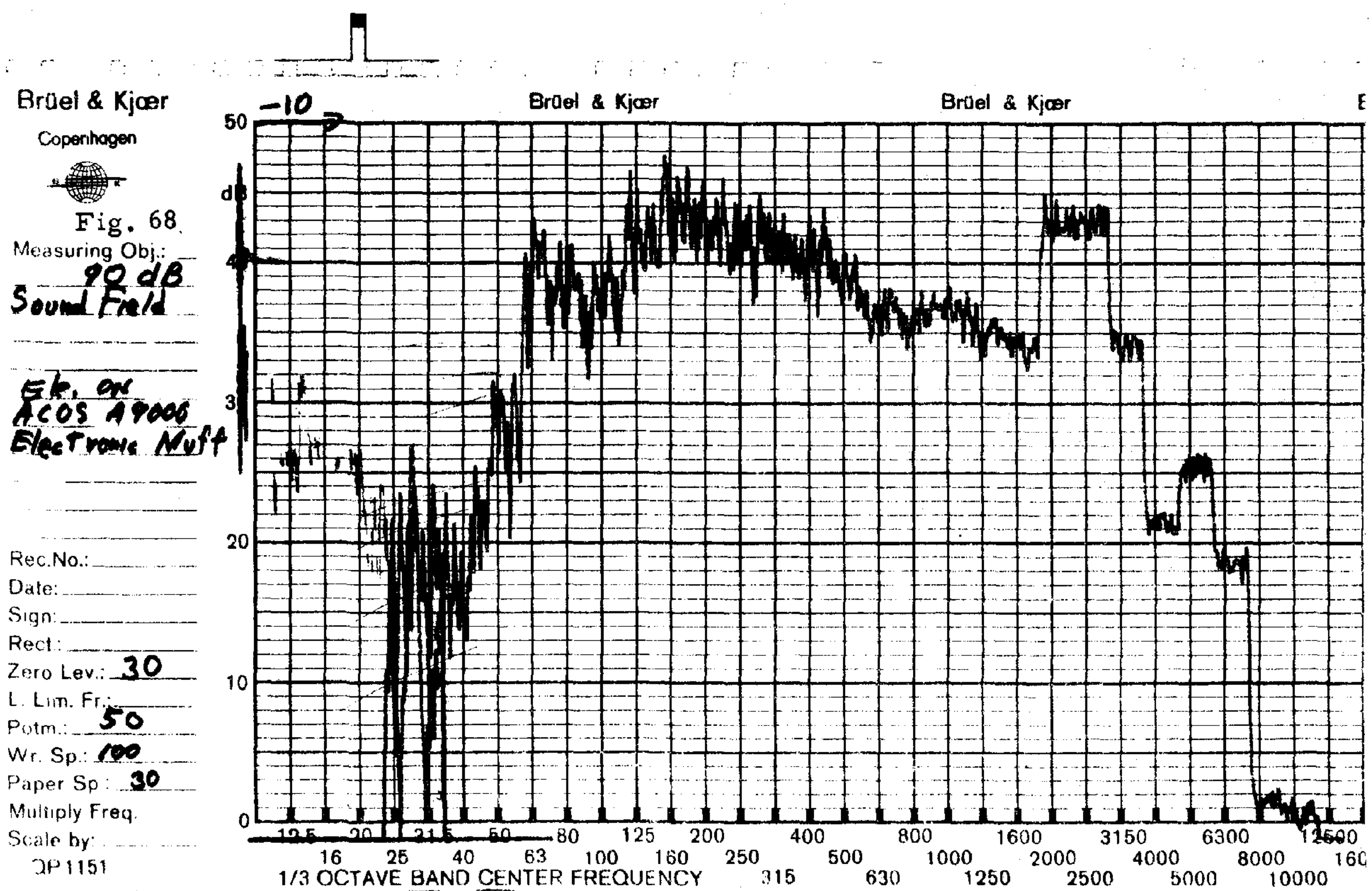

Brüel & Kjær
Copenhagen
Measuring Obj.:
90 dB
Sound Field
Ext. ON
ACOS A 9000
Electronic Muff
Rec.No.:
Date:
Sign.
Rect.:
Zero Lev.: 30
L. Lim. Fr.
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
QP 1151
-10
50
dB
20
10
0
12.5 16 20 25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 800 1000 1250 1600 2000 2500 3150 4000 5000 6300 8000 10000 12500
1/3 OCTAVE BAND CENTER FREQUENCY

Fig. 68

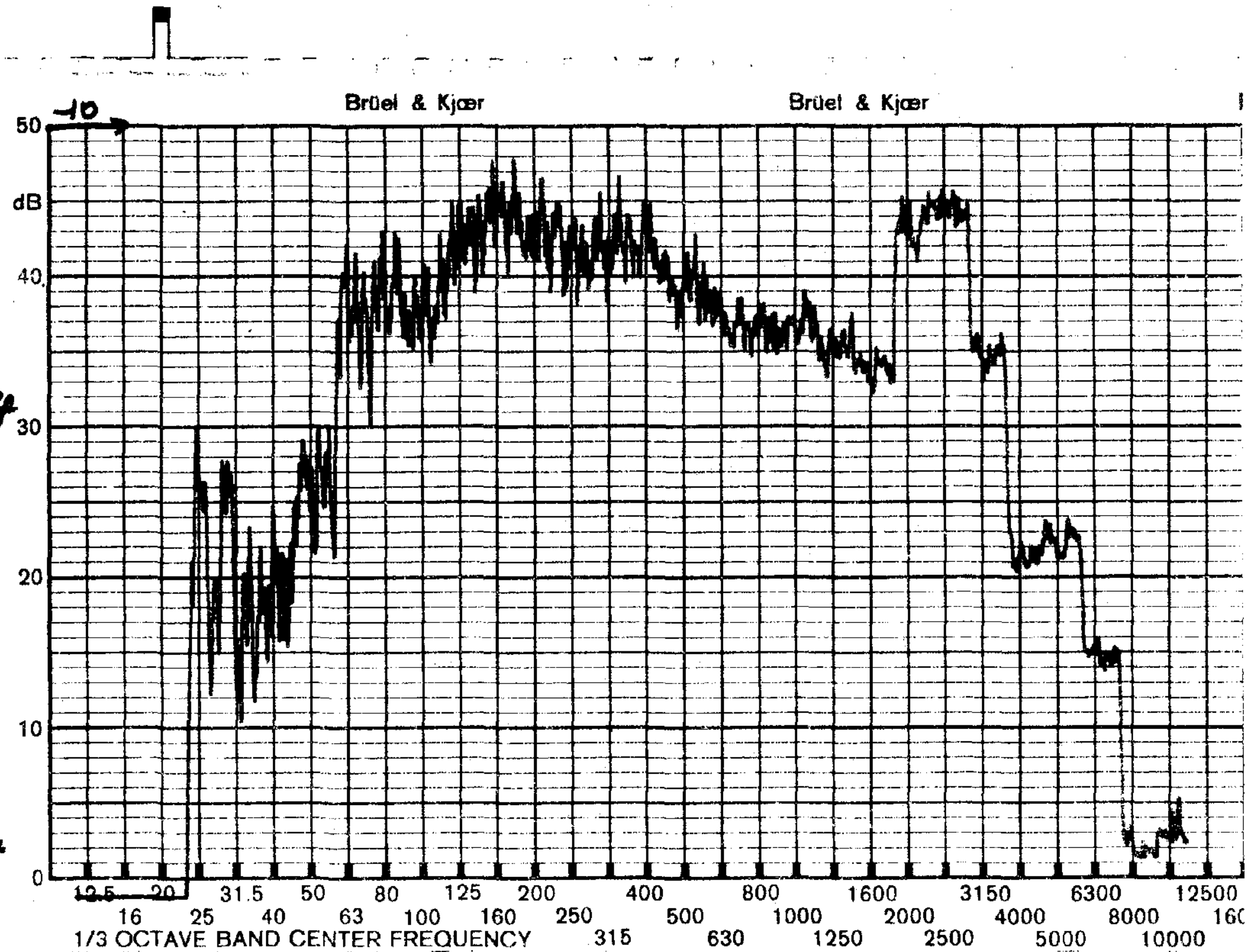
Brüel & Kjær
Copenhagen

Fig. 69

Measuring Obj.: 80 dB Sound Field

ACOS A 9000 Electrone Muff Ele. ON

Rec. No.:
Date:
Sign:
Rect.:
Zero Lev.: 20
L. Lim. Fr.:
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30 mm/sec
Multiply Freq.
Scale by:
QP 1151



Brüel & Kjær
Copenhagen

Fig. 70

Measuring Obj.: ACOS A 9000 Electrone Muff

70 dB Sound Field

Ele. ON

Rec. No.:
Date:
Sign:
Rect.:
Zero Lev.: 20
L. Lim. Fr.:
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
QP 1151

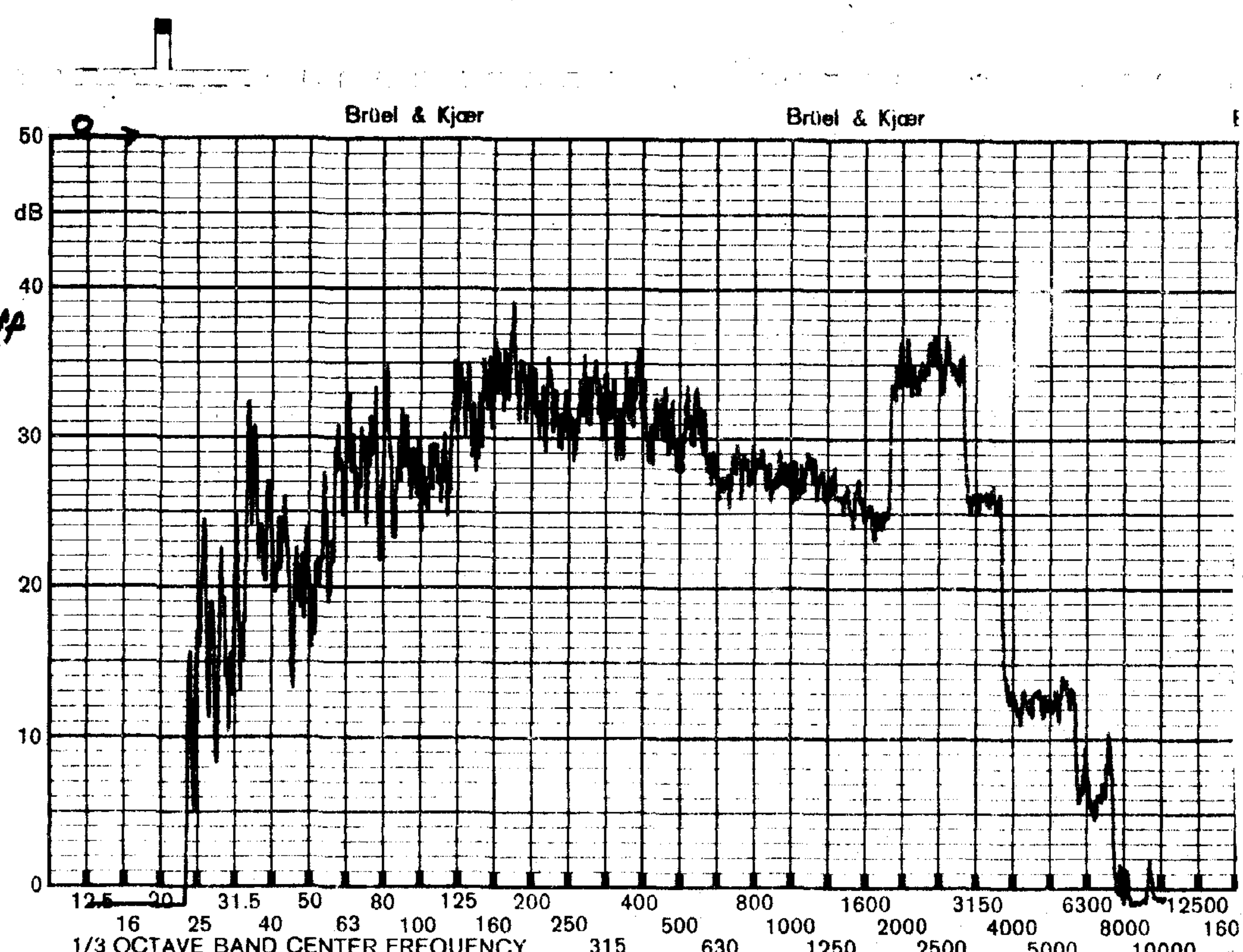
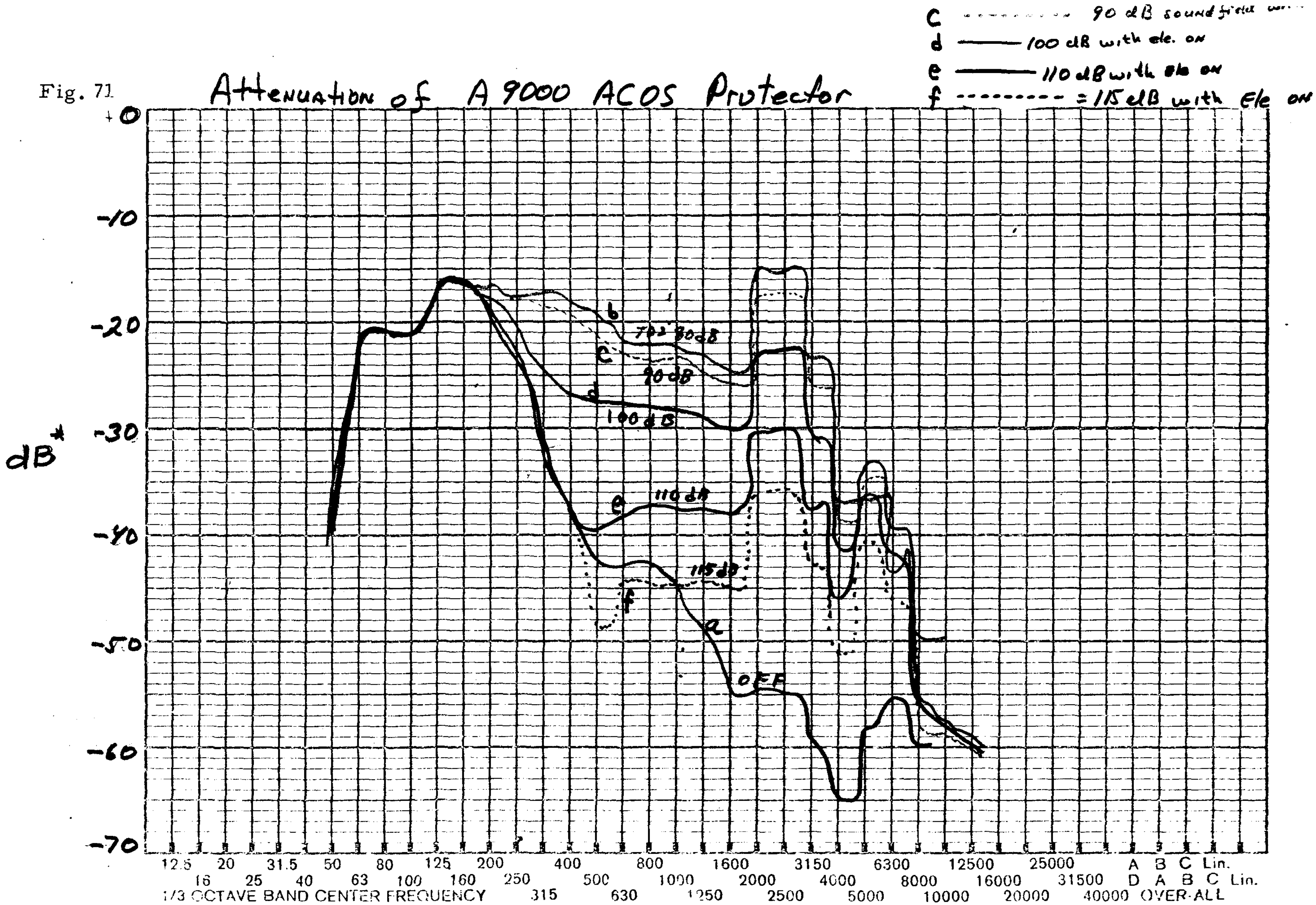


Fig. 71

Attenuation of A 9000 ACOS Protector



* Sound field measured in dB per 63 Hz to 8000 Hz

Add +12 dB to get dB per 1/3 octave.

Brüel & Kjær ← same Brüel & Kjær Brüel & Kjær

Brüel & Kjær
Copenhagen

Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm.

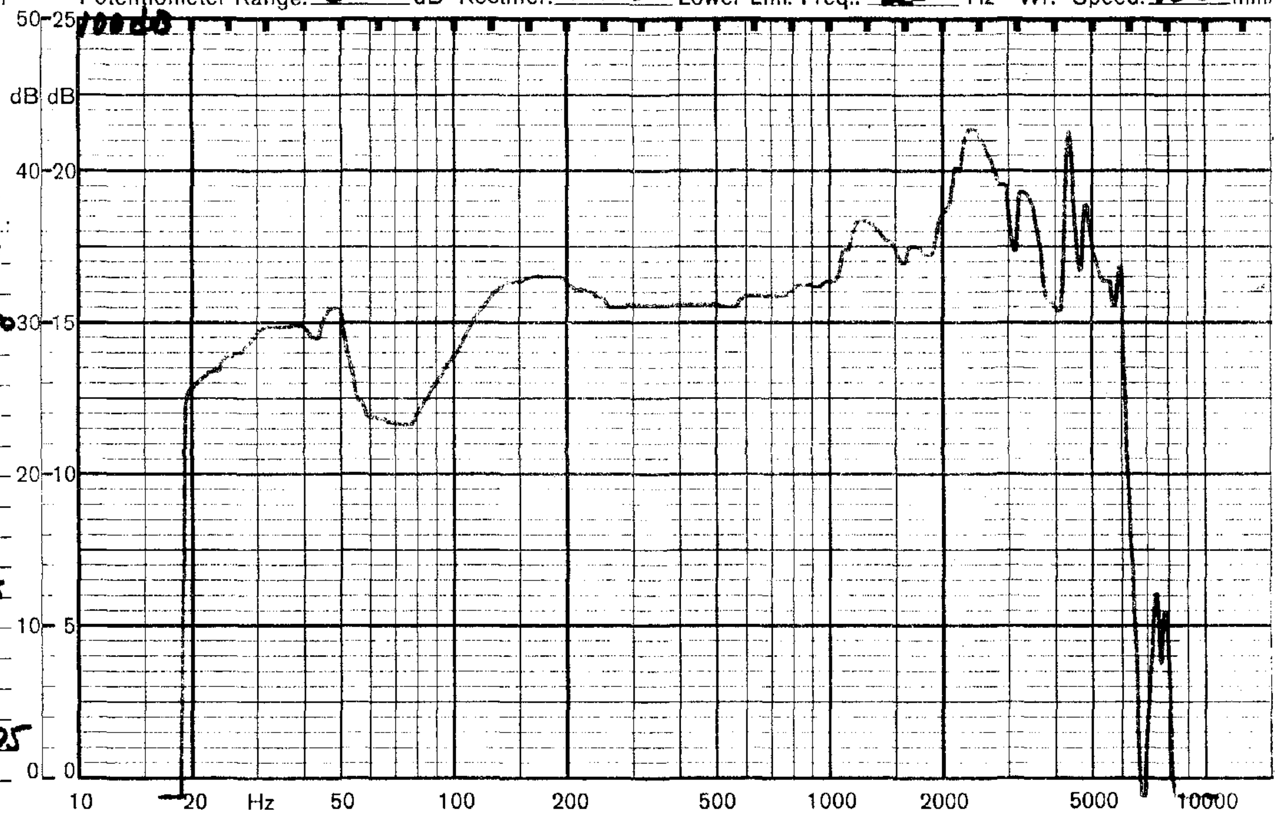


Fig. 72R

Measuring Obj.: ACOS 9000 Right Side Attenuation 100 dB Input

A9000 electronics in box

Rec. No.:
Date: 4-22-75
Sign.: [illegible]

QP 1124 Multiply Frequency Scale by Zero Level: 50 (161

Brüel & Kjær Brüel & Kjær Brüel &

Brüel & Kjær
Copenhagen

Potentiometer Range: 50 dB Rectifier: RMS Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm.

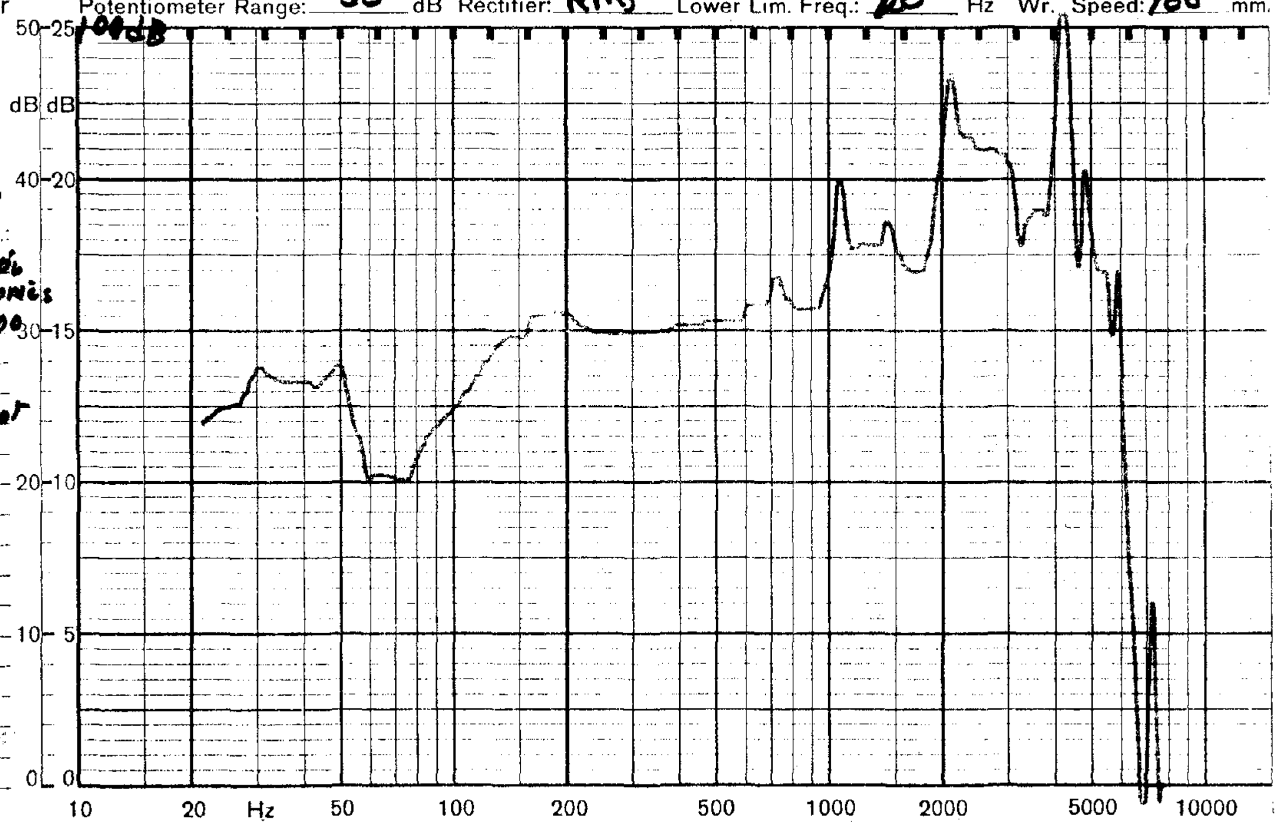


Fig. 72 L

Measuring Obj.: Attenuation of Electronics ACOS A9000 Left Side

100dB input

Rec. No.:
Date:
Sign.:

QP 1124 Multiply Frequency Scale by Zero Level: 50 (161

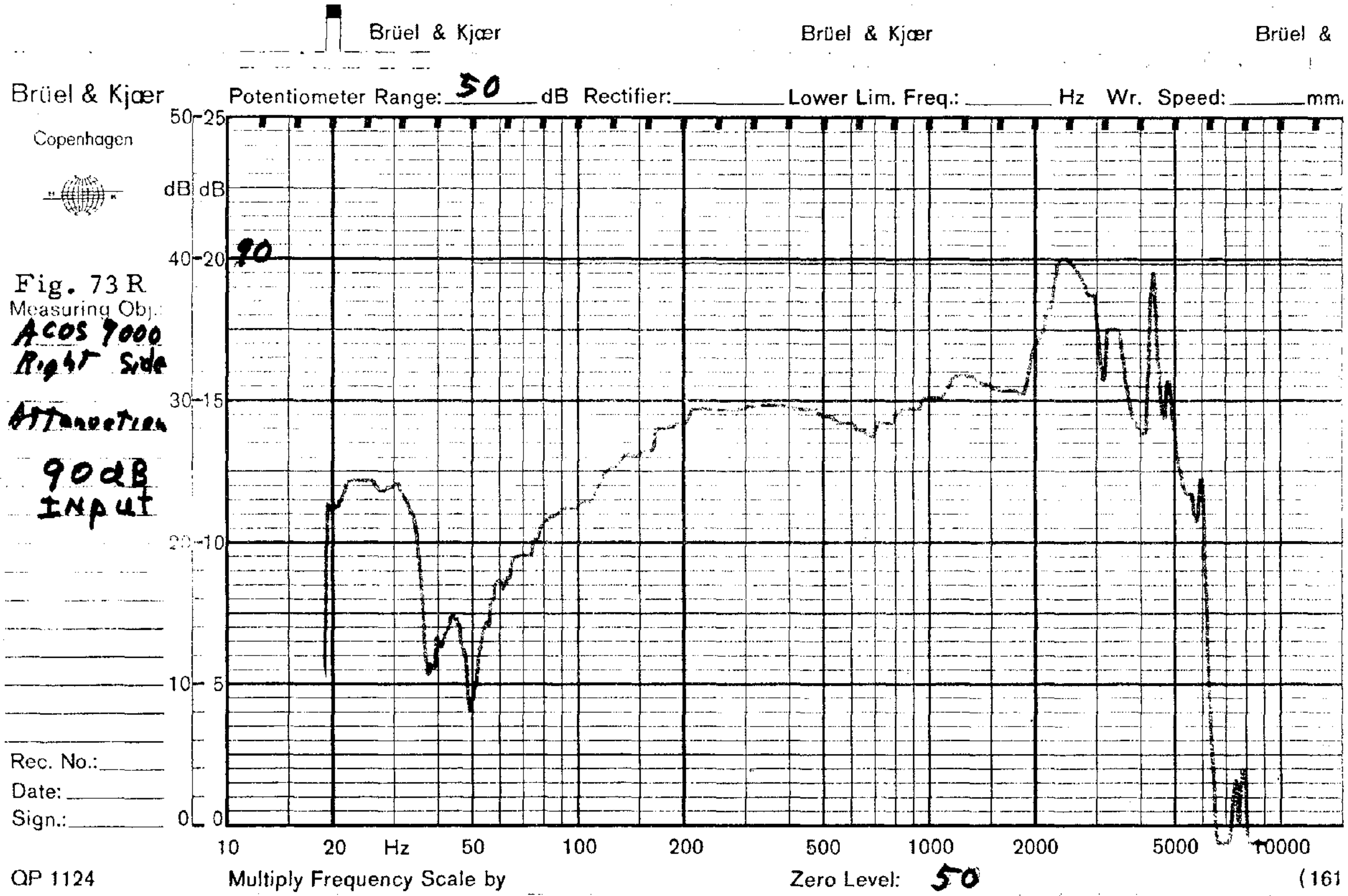


Fig. 73 R

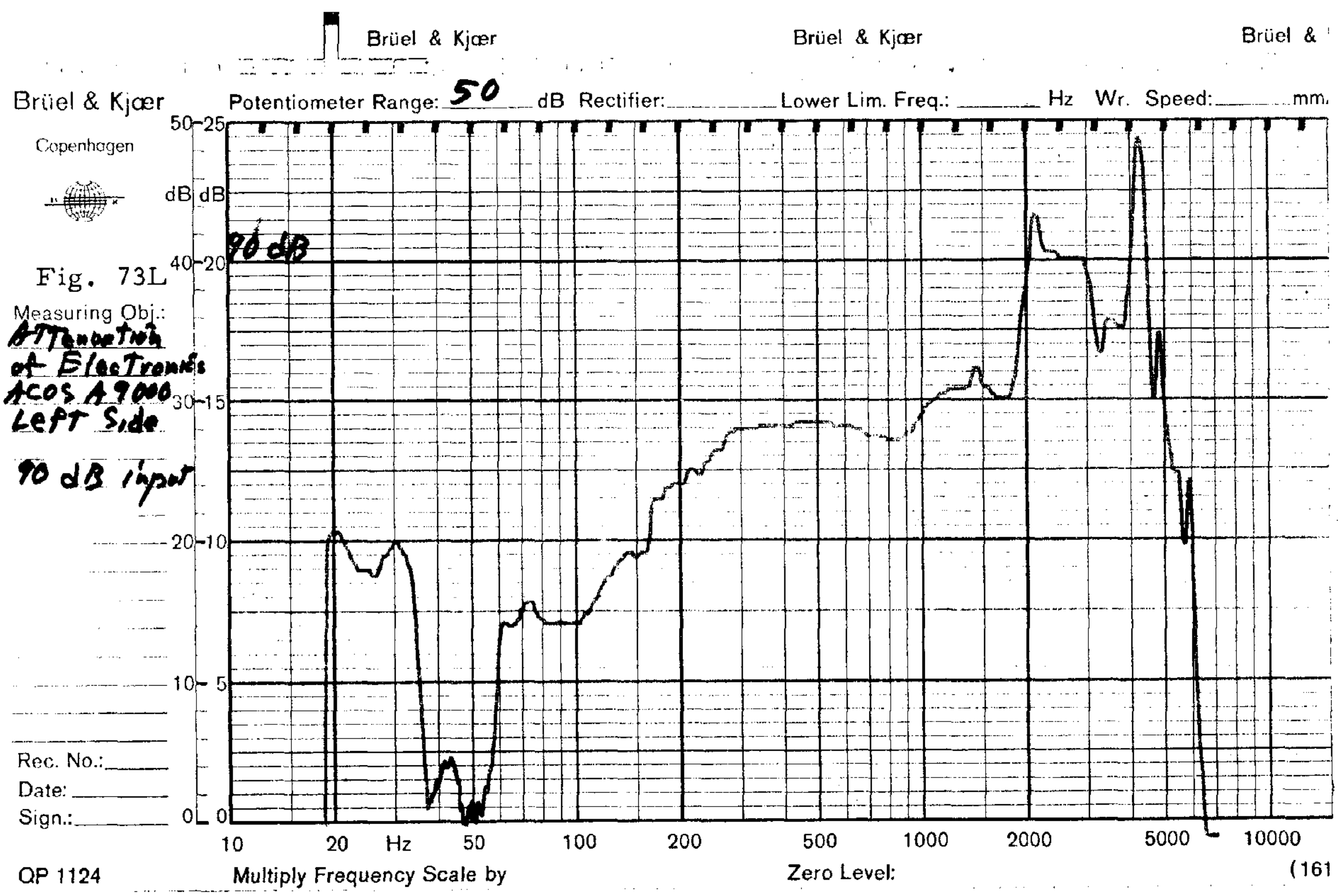
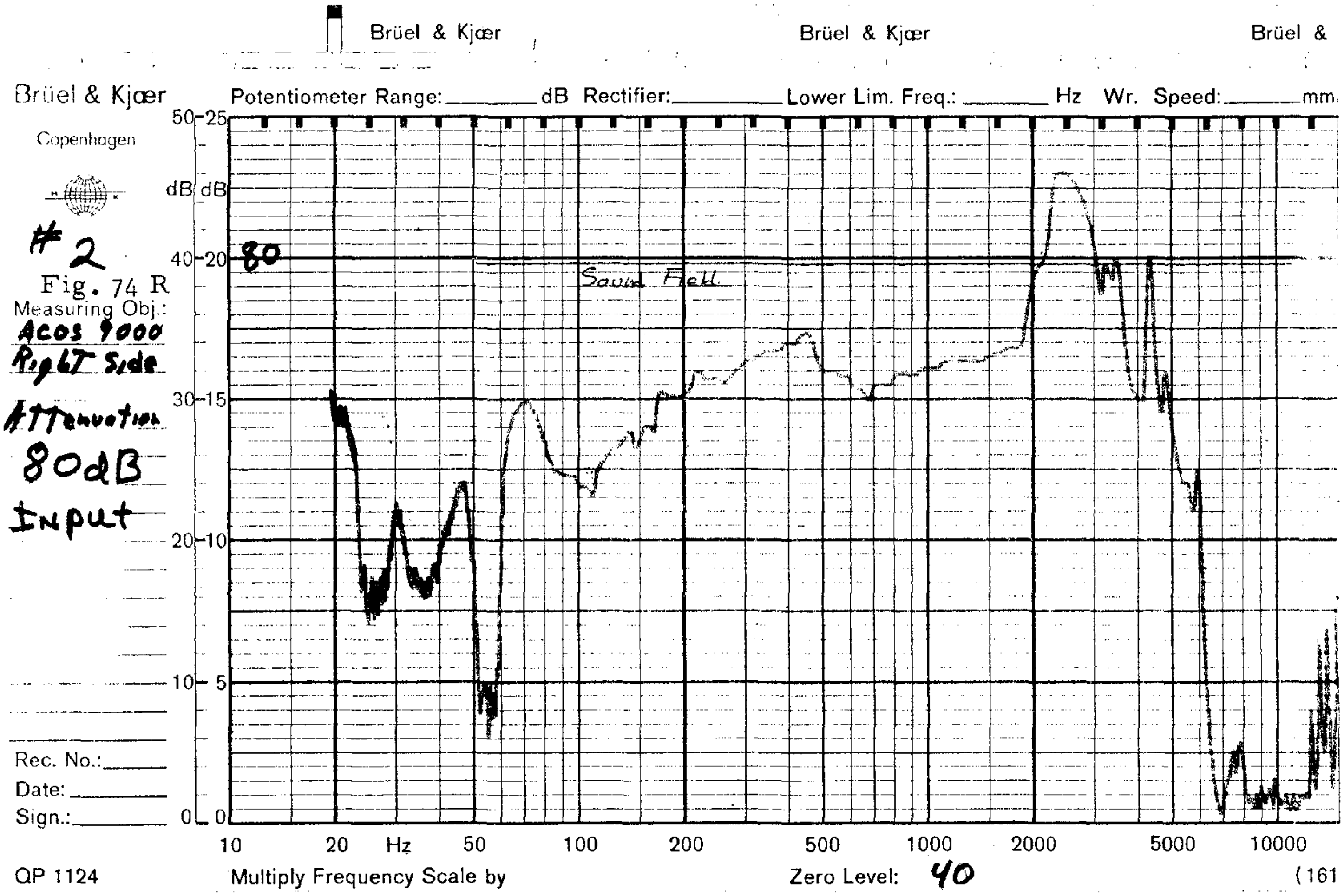
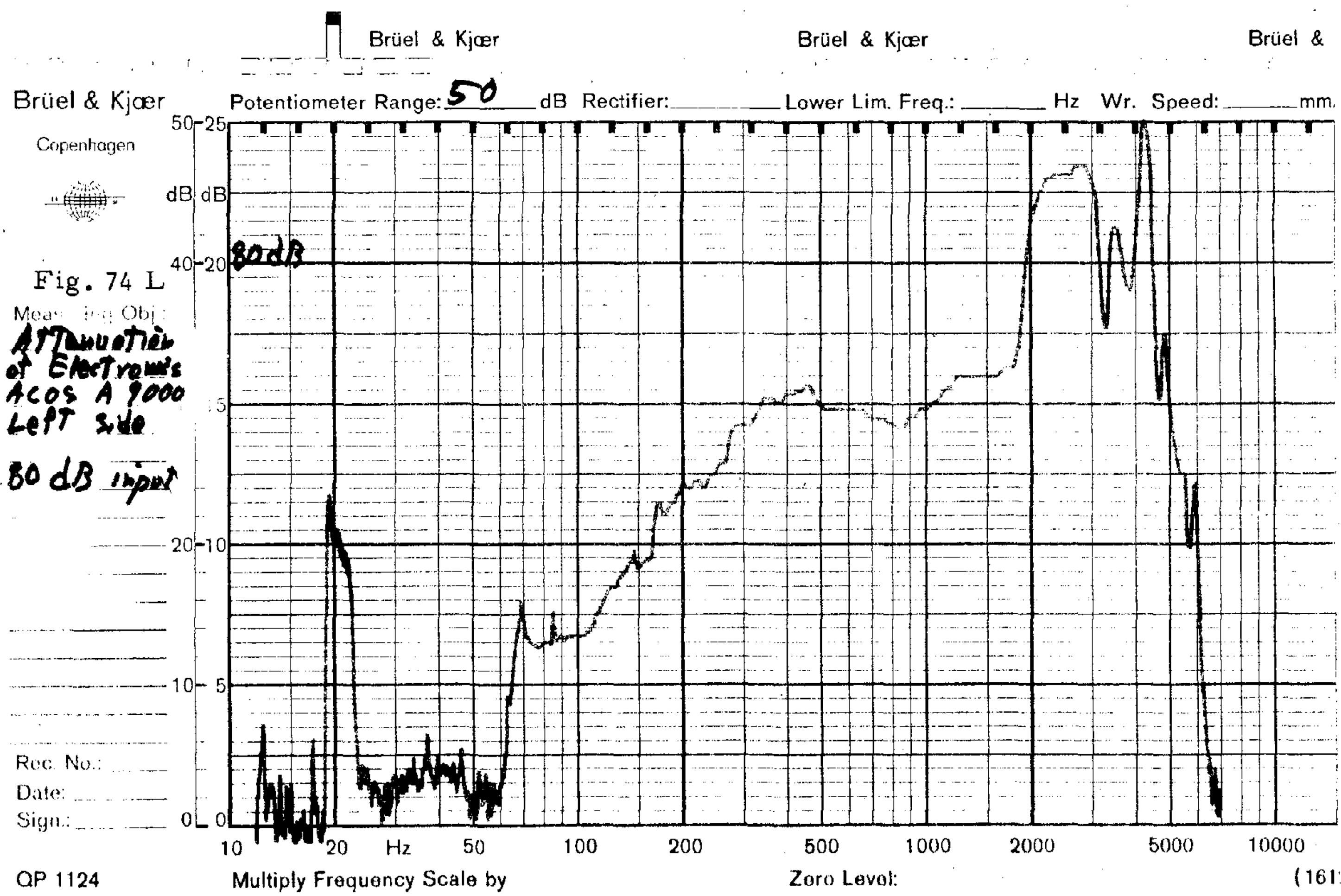


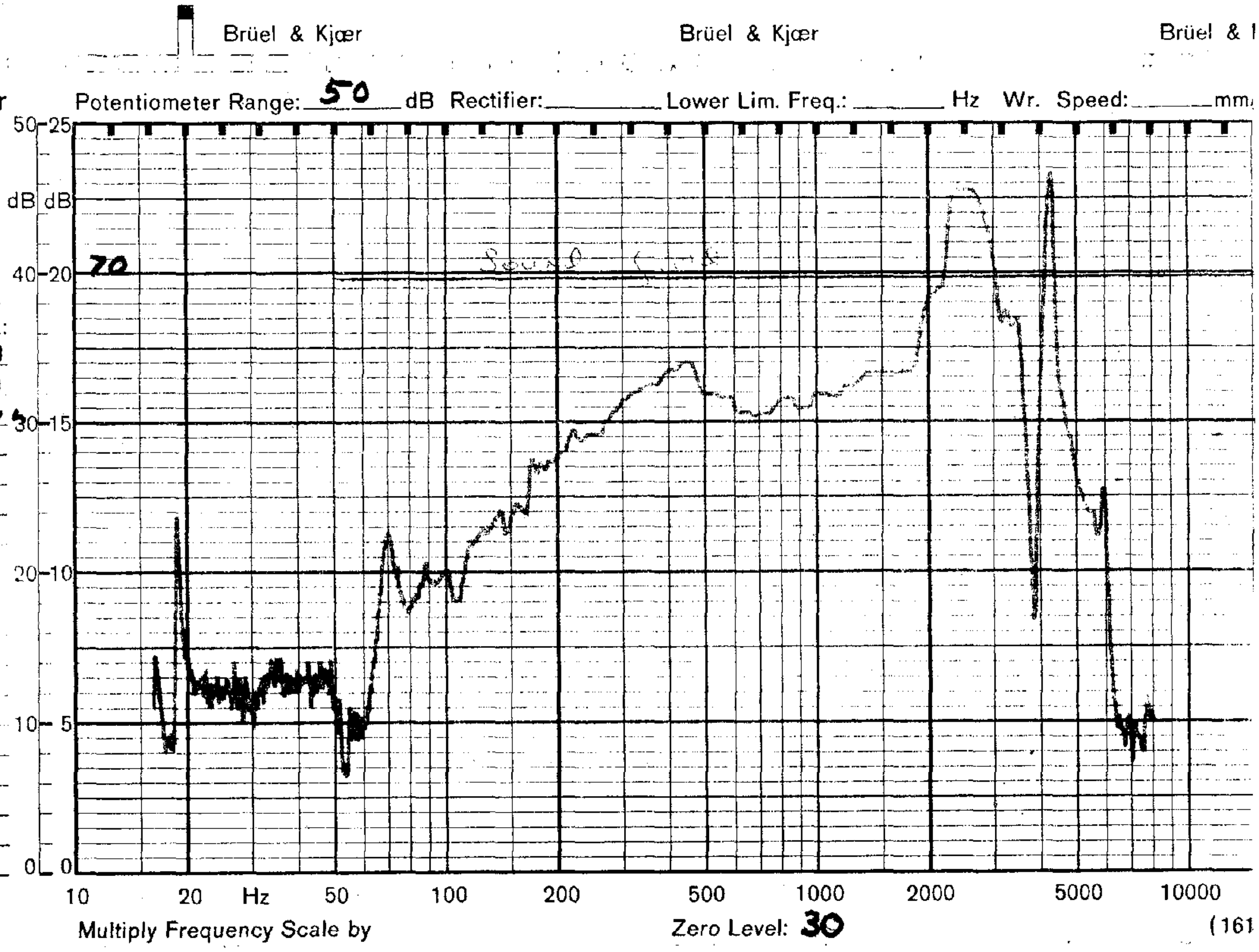
Fig. 73L


Brüel & Kjær
Brüel & Kjær
Brüel &
Brüel & Kjær
Copenhagen
Potentiometer Range: dB Rectifier: Lower Lim. Freq.: Hz Wr. Speed: mm.
2
Fig. 74 R
Measuring Obj.:
ACOS 9000
Right Side
Attenuation
80dB
Input
80
Sound Field
50 25
dB dB
40 20
30 15
20 10
10 5
0 0
Rec. No.:
Date:
Sign.:
10
20
Hz
50
100
200
500
1000
2000
5000
10000
QP 1124
Multiply Frequency Scale by
Zero Level: 40
(161


Brüel & Kjær
Brüel & Kjær
Brüel &
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: Lower Lim. Freq.: Hz Wr. Speed: mm.
Fig. 74 L
Attenuation of Electronics
ACOS A 9000
Left Side
80 dB input
80dB
50 25
dB dB
40 20
20 10
10 5
0 0
Rec. No.:
Date:
Sign.:
10
20
Hz
50
100
200
500
1000
2000
5000
10000
QP 1124
Multiply Frequency Scale by
Zero Level:
(161

Brüel & Kjær
Copenhagen

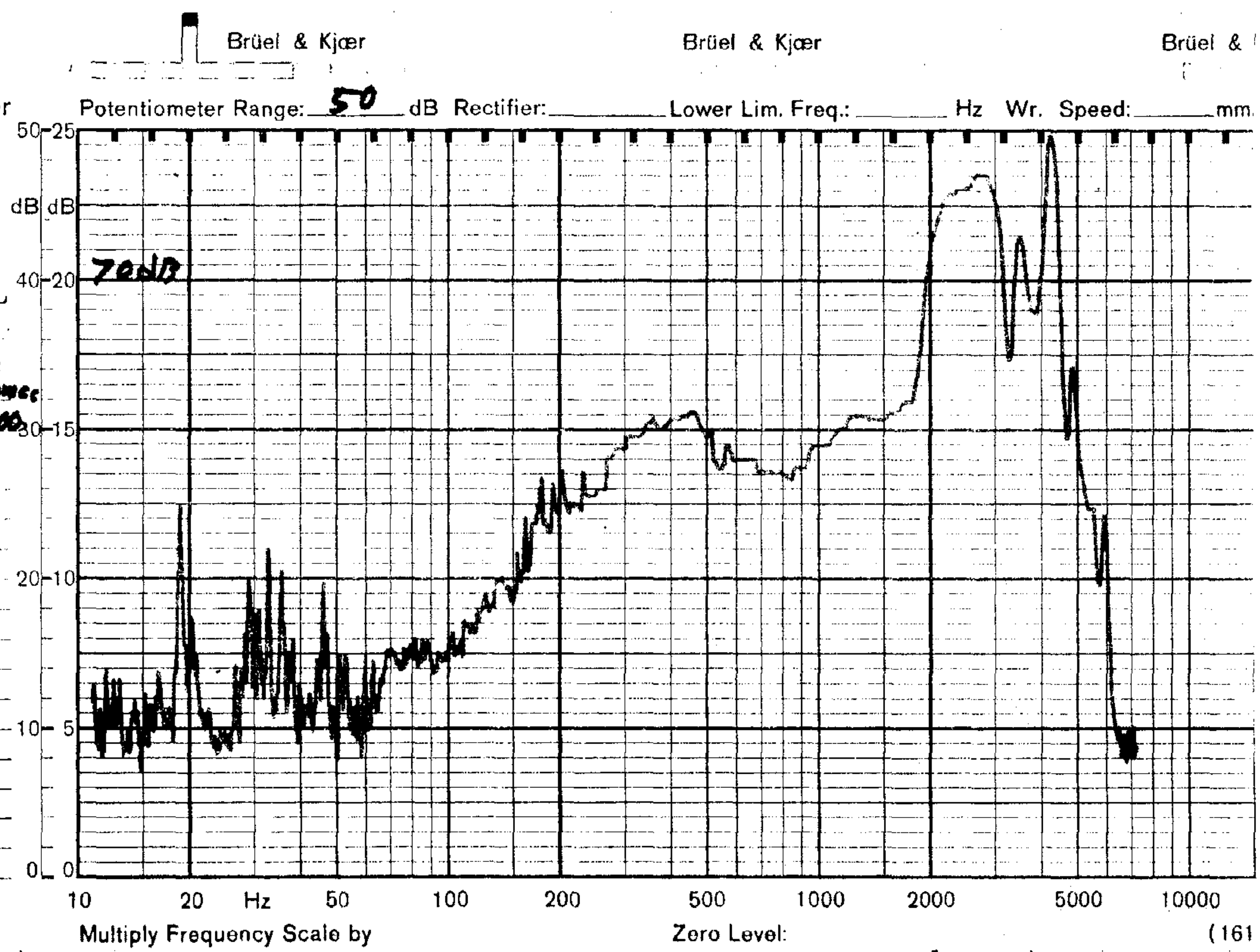
Fig. 75 R
Measuring Obj.:
ACOS 9000
Right Side
Attenuation
Input
70 dB



Rec. No.:
Date:
Sign.:
QP 1124

Brüel & Kjær
Copenhagen

Fig. 75 L
Measuring Obj.:
Attenuation of Electronic ACOS A 9000 ~~Right~~ Left Side
70 dB Input



Rec. No.:
Date:
Sign.:
QP 1124

Fig. 76R

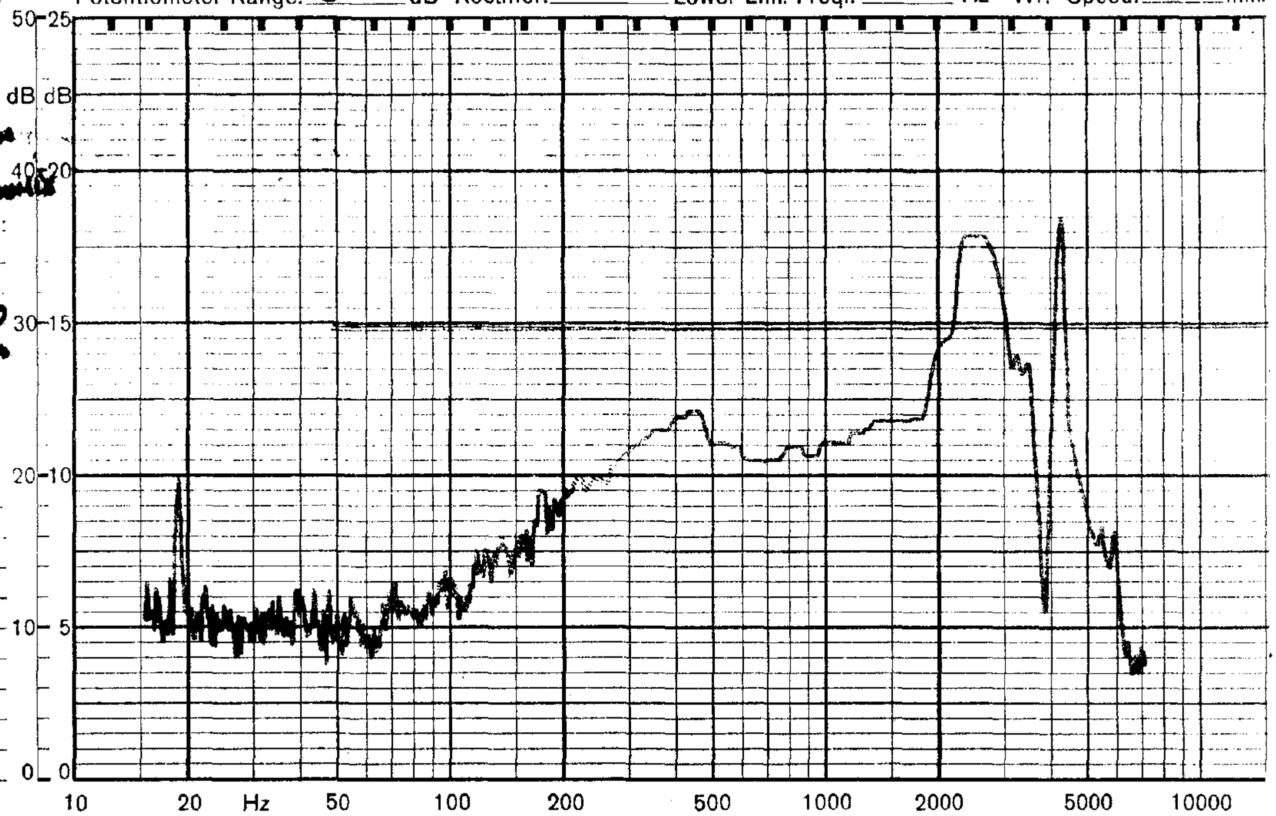
Brüel & Kjær Copenhagen

Potentiometer Range: 50 dB Rectifier: Lower Lim. Freq.: Hz Wr. Speed: mm/

plots of attenuation of Acos 9000 electronics

Measuring Obj.: Acos 9000 Right Side 60 Attenuation

60 dB



Rec. No.:
Date:
Sign.:

QP 1124 Multiply Frequency Scale by Zero Level: 30 (161

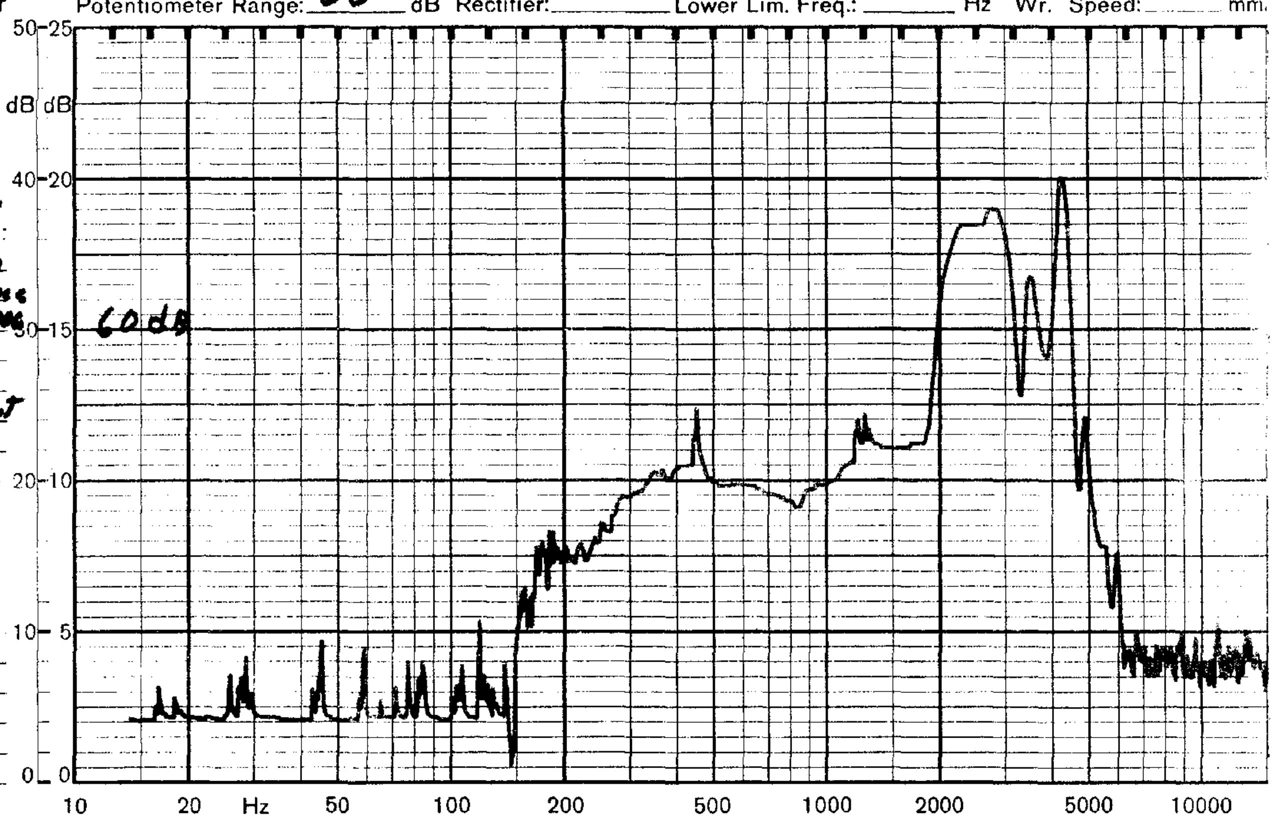
Fig. 76 L

Brüel & Kjær Copenhagen

Potentiometer Range: 50 dB Rectifier: Lower Lim. Freq.: Hz Wr. Speed: mm.

Measuring Obj.: Attenuation of Electronic ACOS A-9000 Left side

60 dB input



Rec. No.:
Date:
Sign.:

QP 1124 Multiply Frequency Scale by Zero Level: (161

Fig. 77

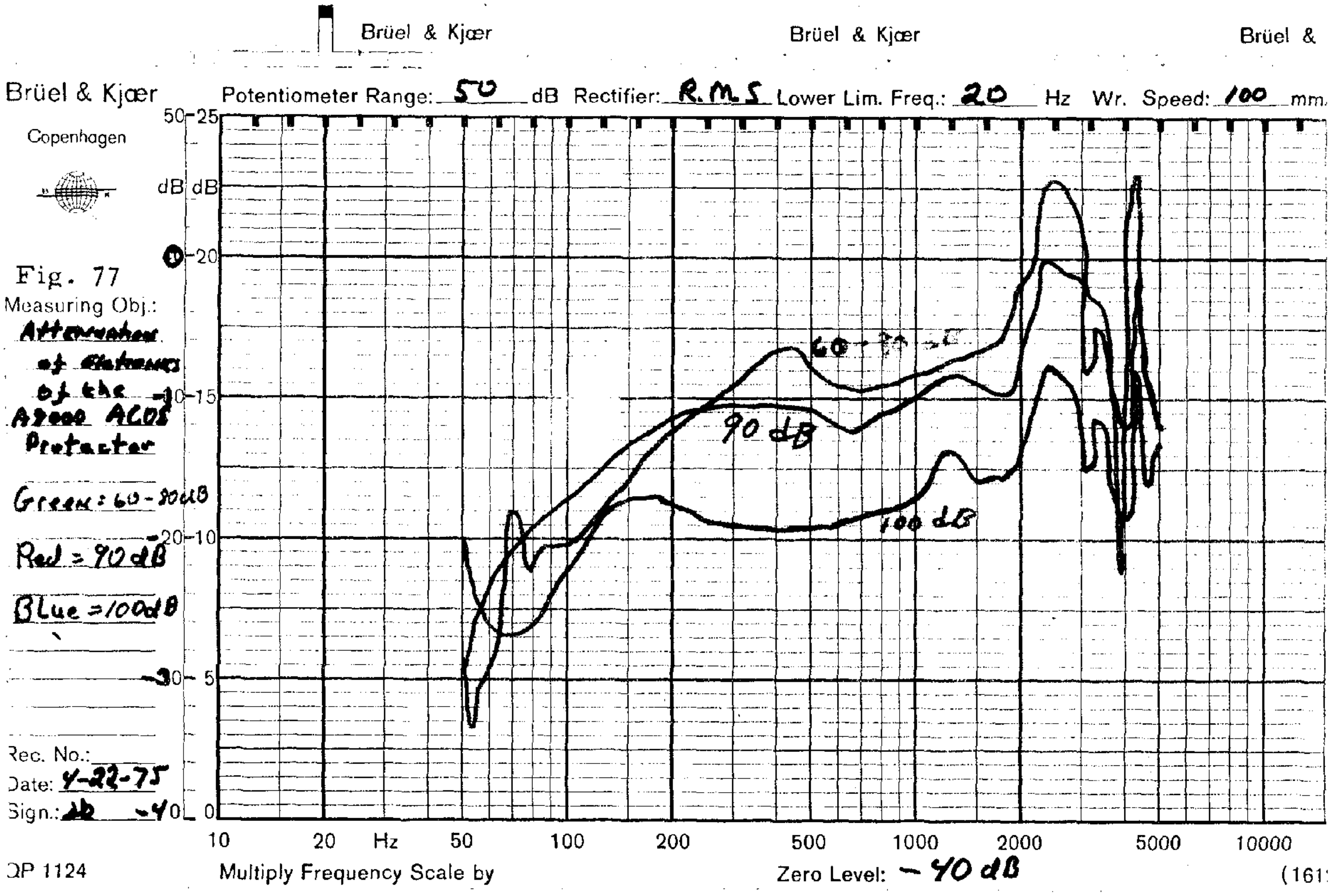
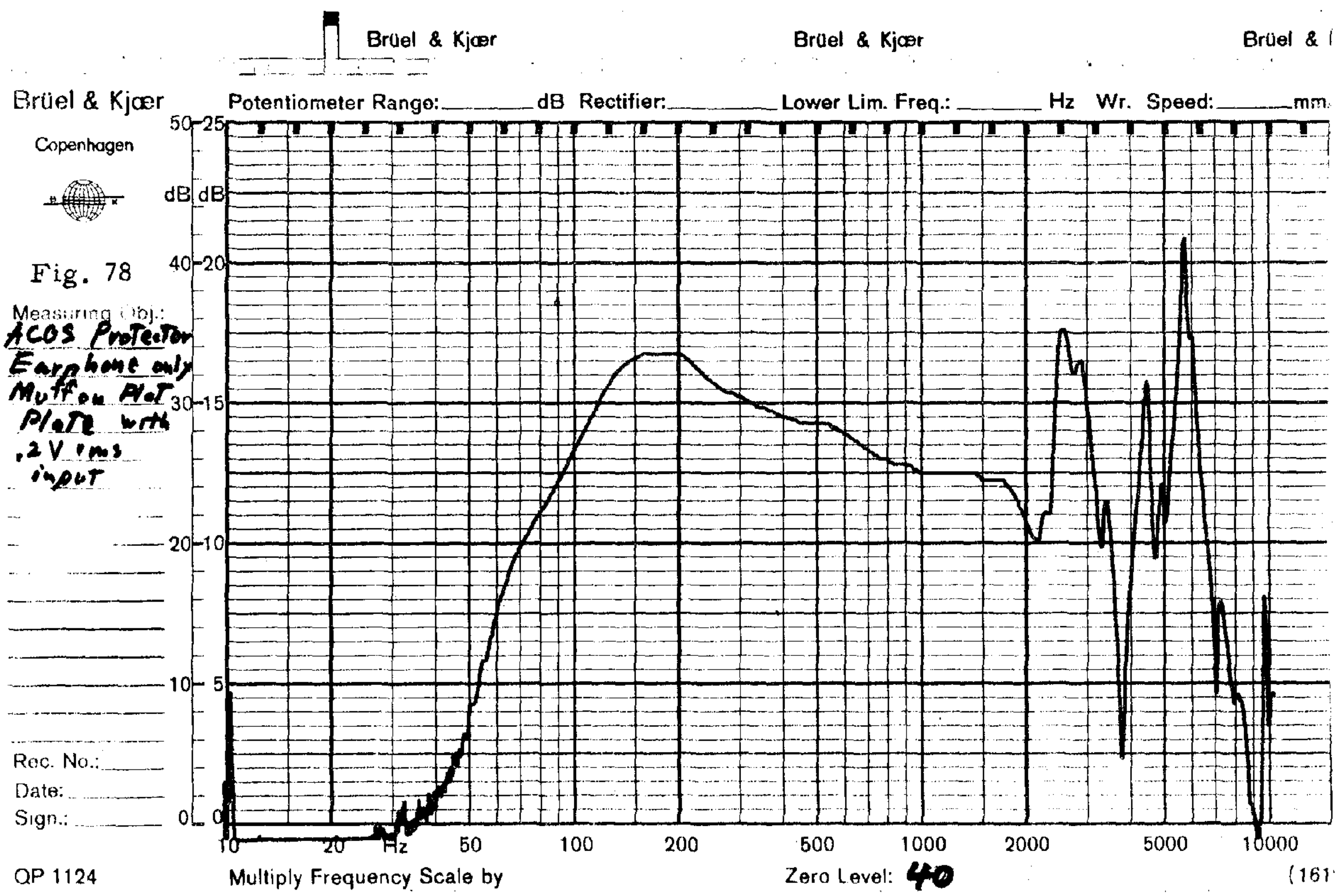

Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: R.M.S Lower Lim. Freq.: 20 Hz Wr. Speed: 100 mm.
Measuring Obj.:
Green: 60-80 dB
Red = 90 dB
Blue = 100 dB
90 dB
100 dB
Rec. No.:
Date: 4-22-75
Sign.:
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: -40 dB

Fig. 78


Brüel & Kjær
Copenhagen
Potentiometer Range: dB Rectifier: Lower Lim. Freq.: Hz Wr. Speed: mm.
Measuring Obj.:
ACOS Protector
Earphone only
Muff on Flat Plate with
.2 V rms input
Rec. No.:
Date:
Sign.:
10 20 Hz 50 100 200 500 1000 2000 5000 10000
QP 1124
Multiply Frequency Scale by
Zero Level: 40

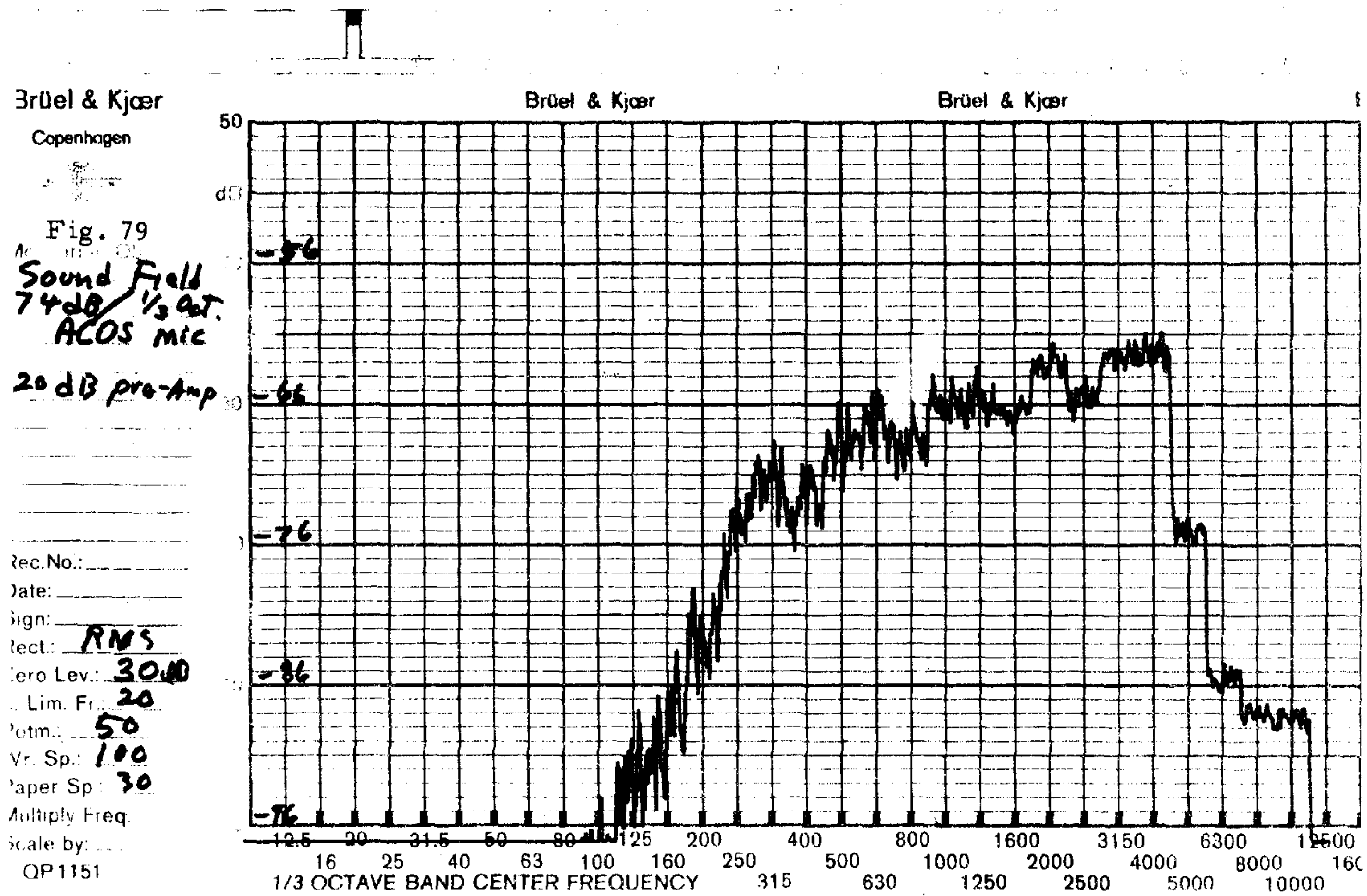

Brüel & Kjær
Copenhagen
Sound Field
74 dB/⅓ Oct.
ACOS mic
20 dB pre-Amp
Rec. No.:
Date:
Sign:
Rect.: RMS
Zero Lev.: 30 dB
L. Lim. Fr.: 20
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
QP 1151
Brüel & Kjær
Brüel & Kjær
50
dB
12.5 16 20 25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 800 1000 1250 1600 2000 2500 3150 4000 5000 6300 8000 10000 12500
1/3 OCTAVE BAND CENTER FREQUENCY

Fig. 79

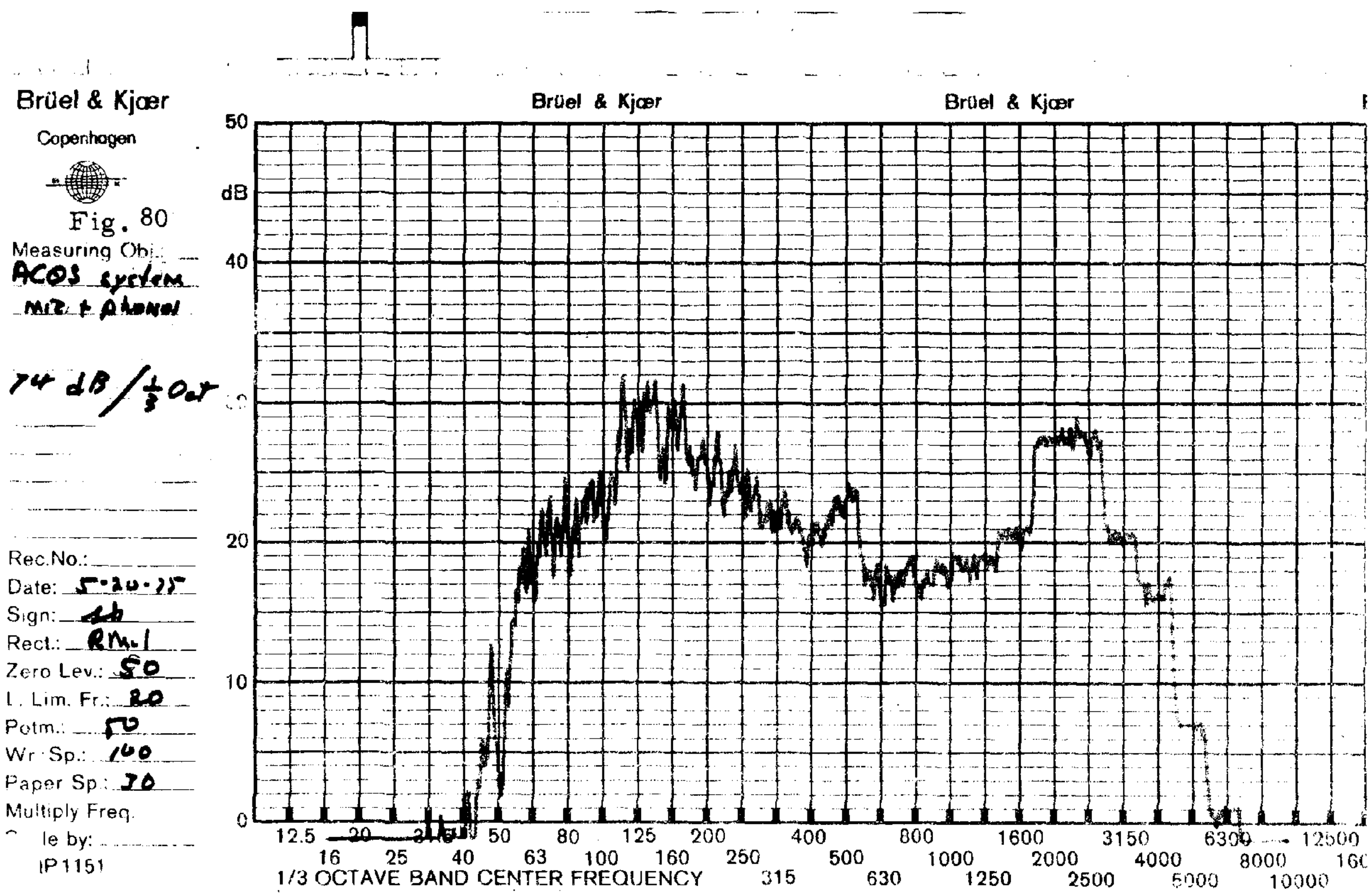

Brüel & Kjær
Copenhagen
Measuring Obj.:
ACOS system
74 dB/⅓ Oct
Rec. No.:
Date: 5-20-75
Sign:
Rect.: RMS
Zero Lev.: 50
L. Lim. Fr.: 20
Potm.: 50
Wr. Sp.: 100
Paper Sp.: 30
Multiply Freq.
Scale by:
QP 1151
Brüel & Kjær
Brüel & Kjær
50
dB
40
20
10
0
12.5 16 20 25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 800 1000 1250 1600 2000 2500 3150 4000 5000 6300 8000 10000 12500
1/3 OCTAVE BAND CENTER FREQUENCY

Fig. 80

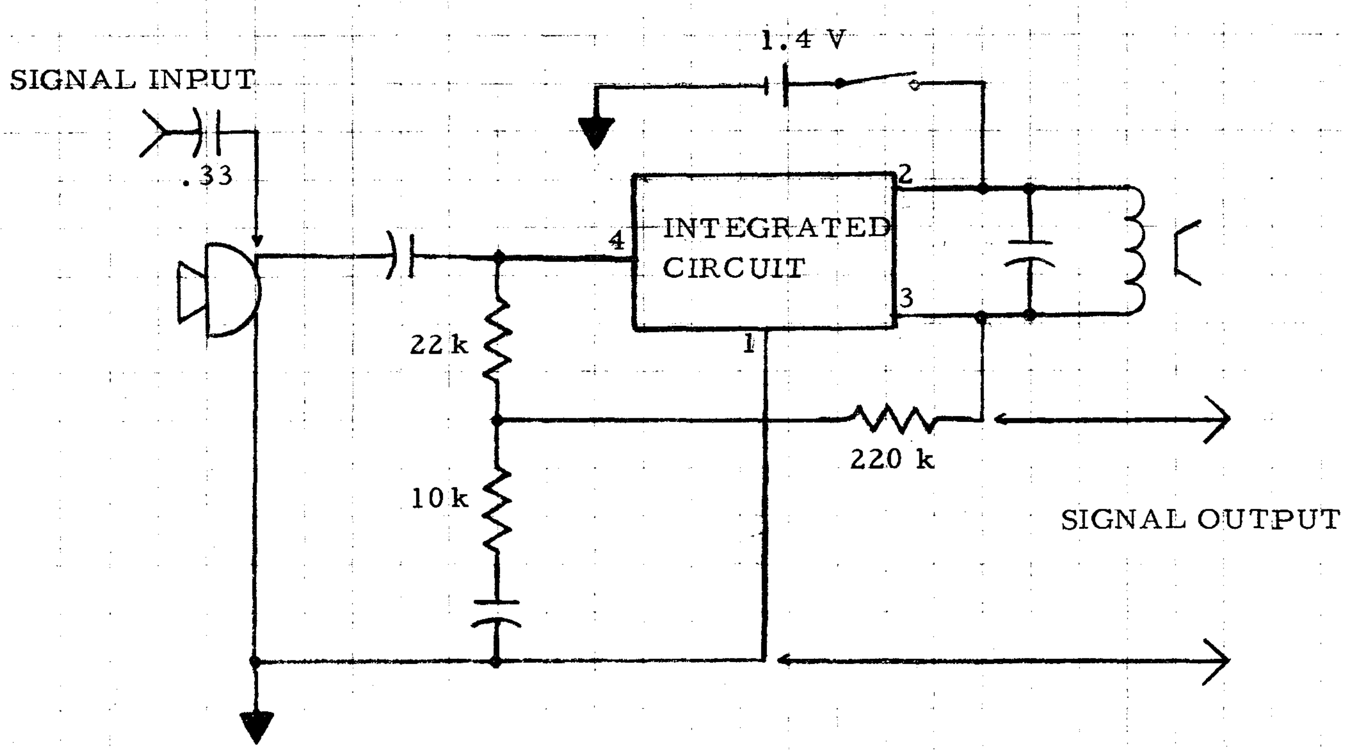
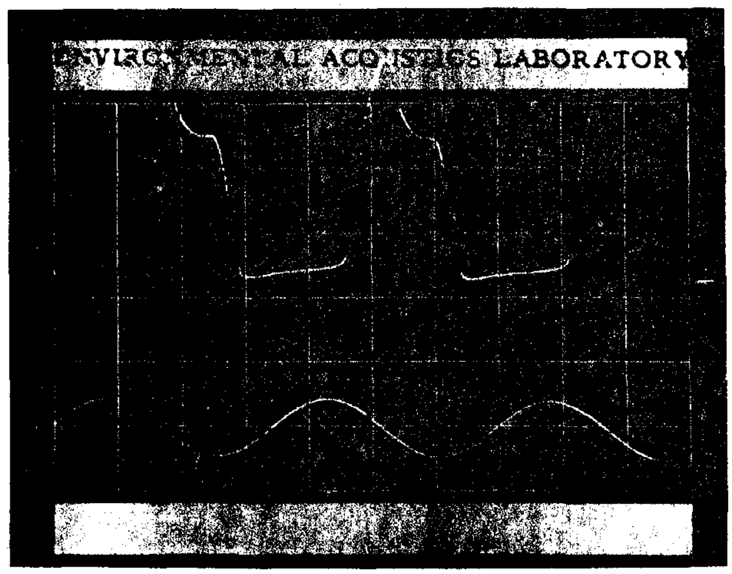
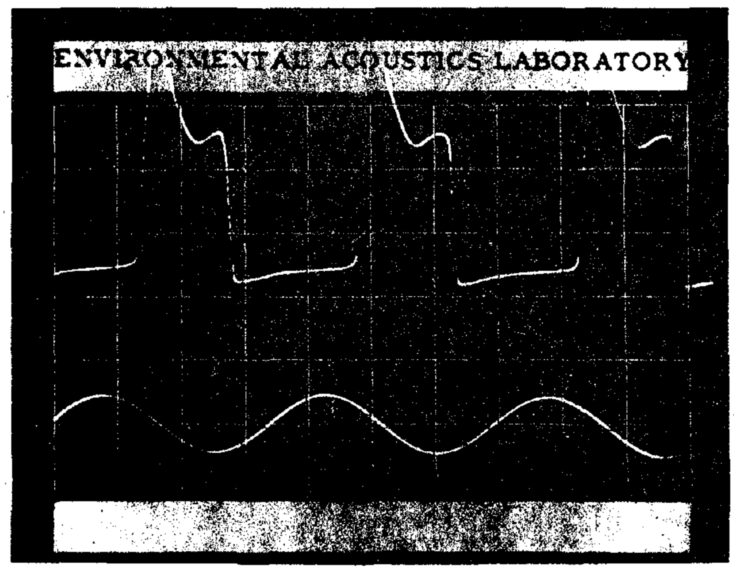


Fig. 81 ACOS A9000/2 Circuit diagram Showing added input and output connections.

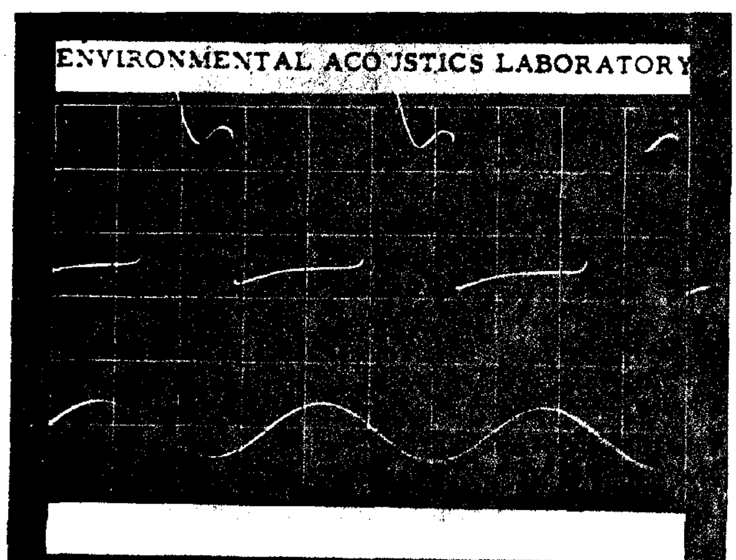
C. Waveform for input e+10dB.



D. Waveform for input e+15dB.



E. Waveform for input e+20dB.



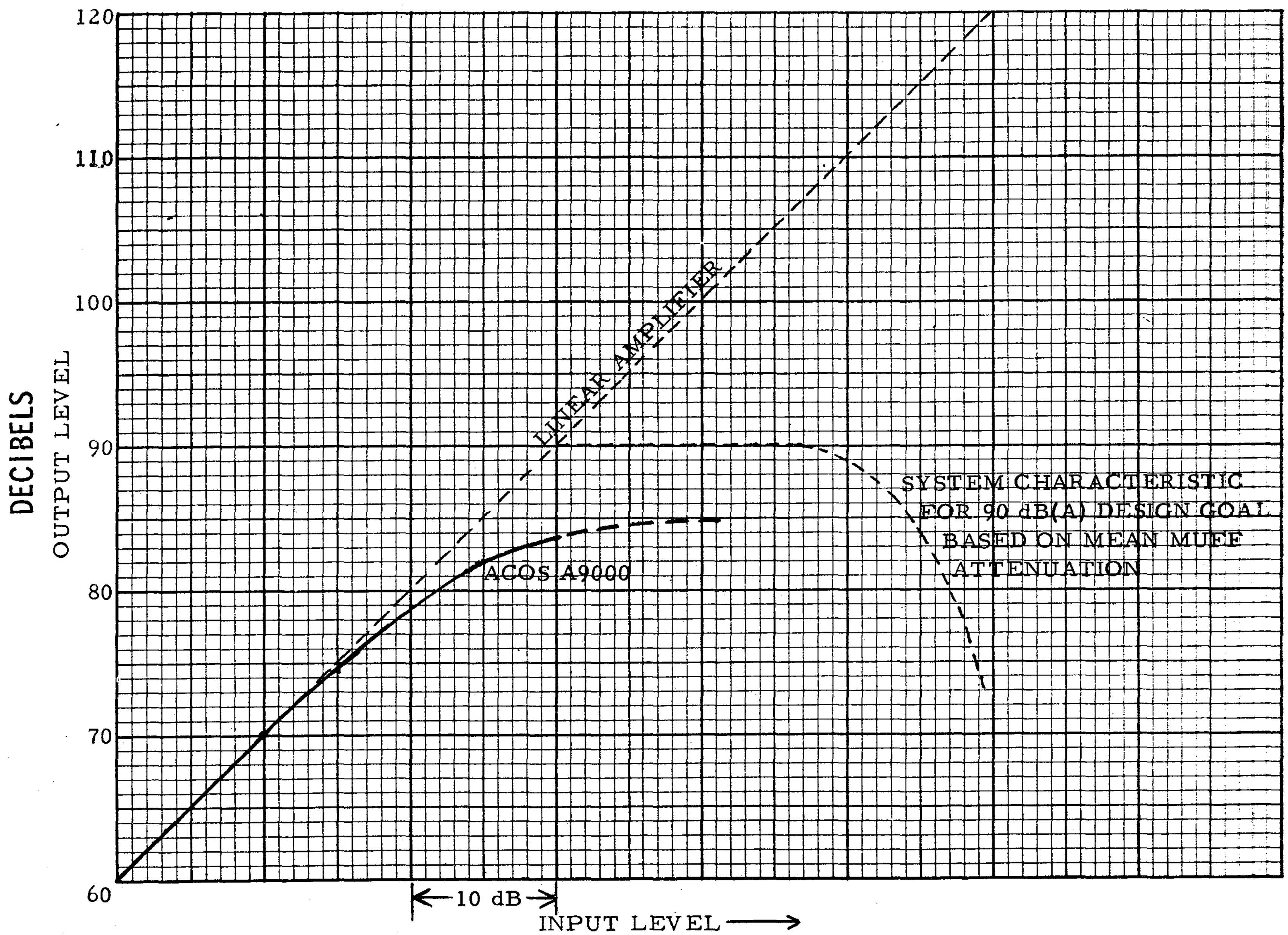


Fig. 83 Transfer characteristics of ACOS A9000 Hearing Protector

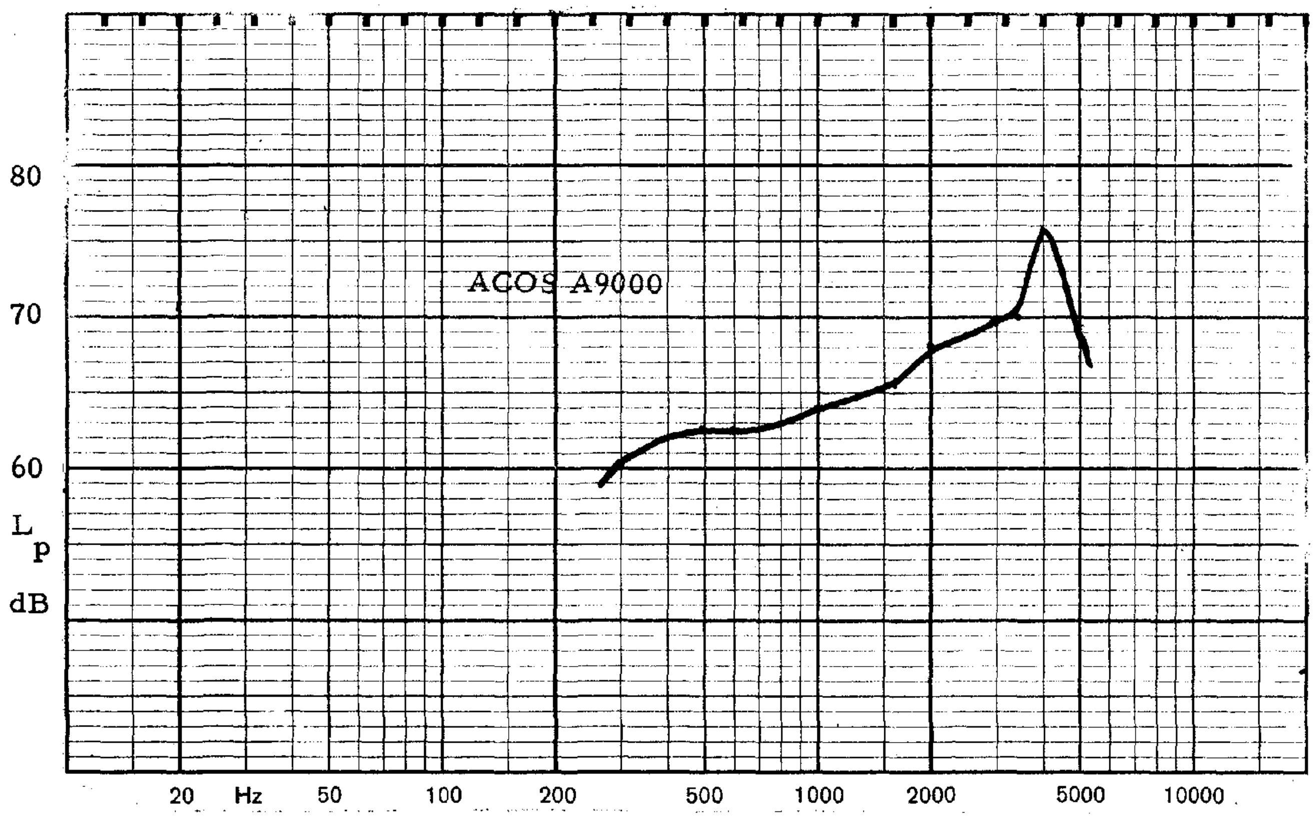


Fig. 84
Frequency response of ACOS A9000 Electronic hearing protector with electrical input injected in parallel with microphone and acoustic output read on B&K 2204 Sound Level Meter with flat plate coupler.

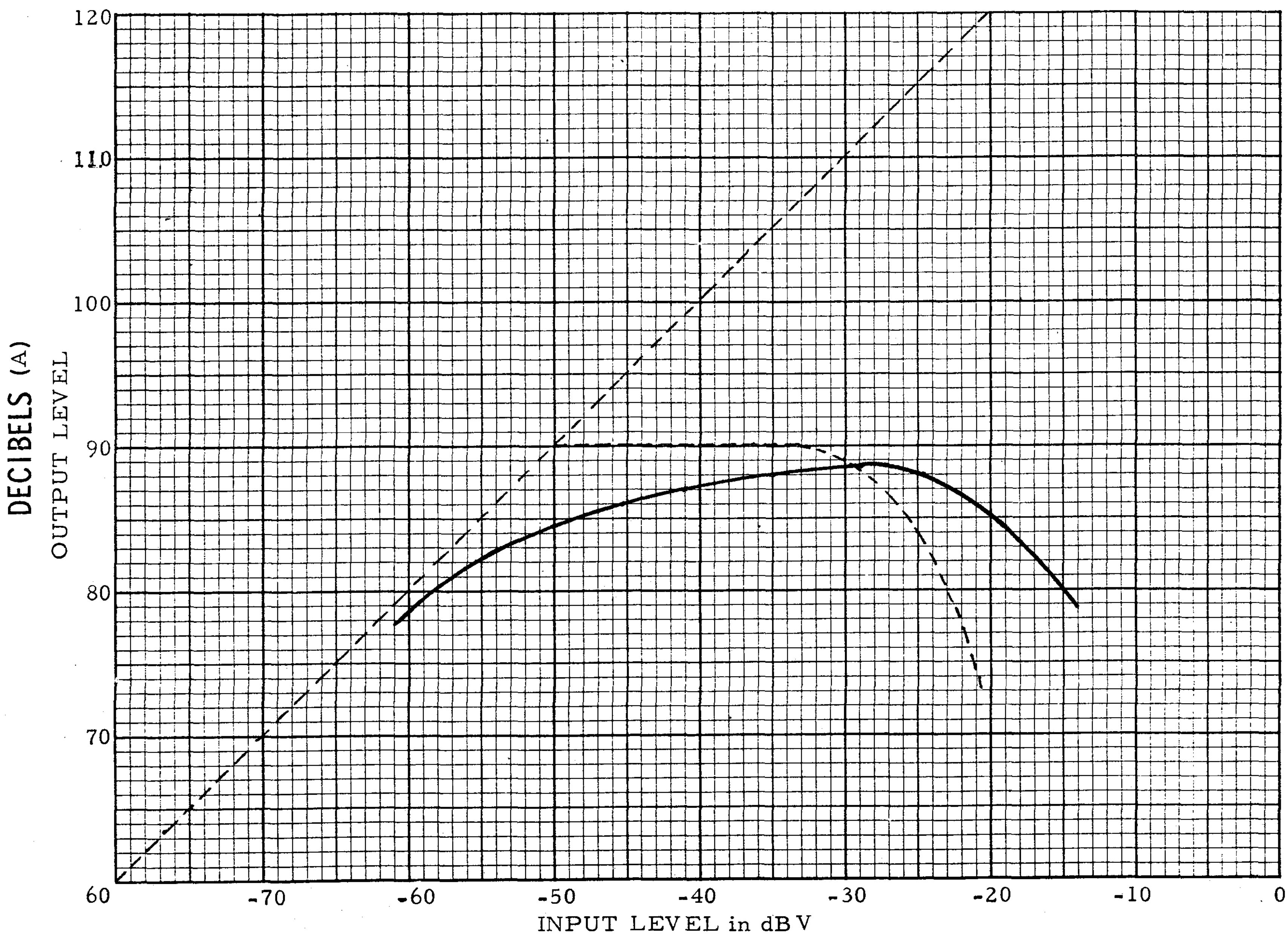


Fig. 85 ACOUSTIC OUTPUT OF ACOS HEARING PROTECTOR FOR PINK NOISE INPUT

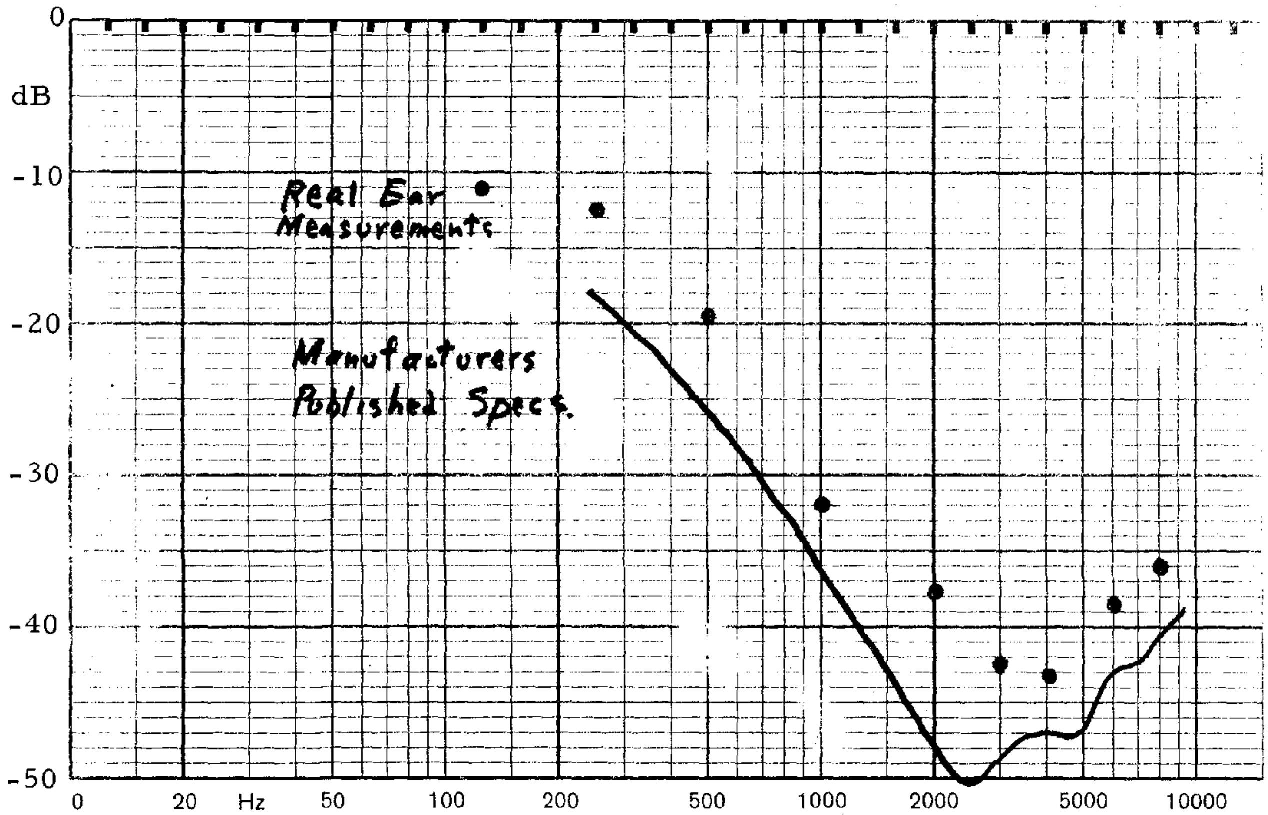


Fig.86 Comparison between real-ear attenuation of ACOS A9000/2 hearing protector and published specifications.

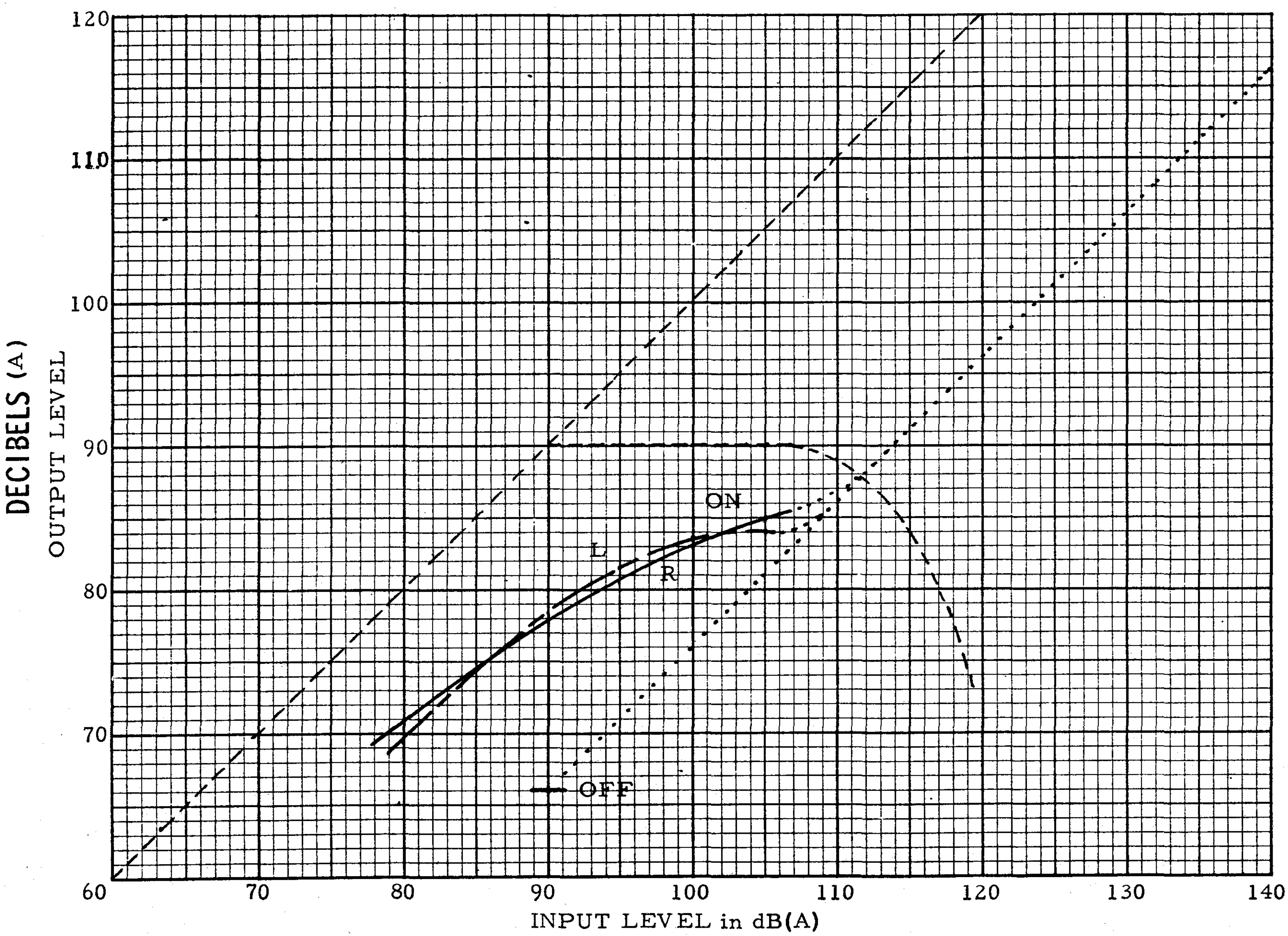
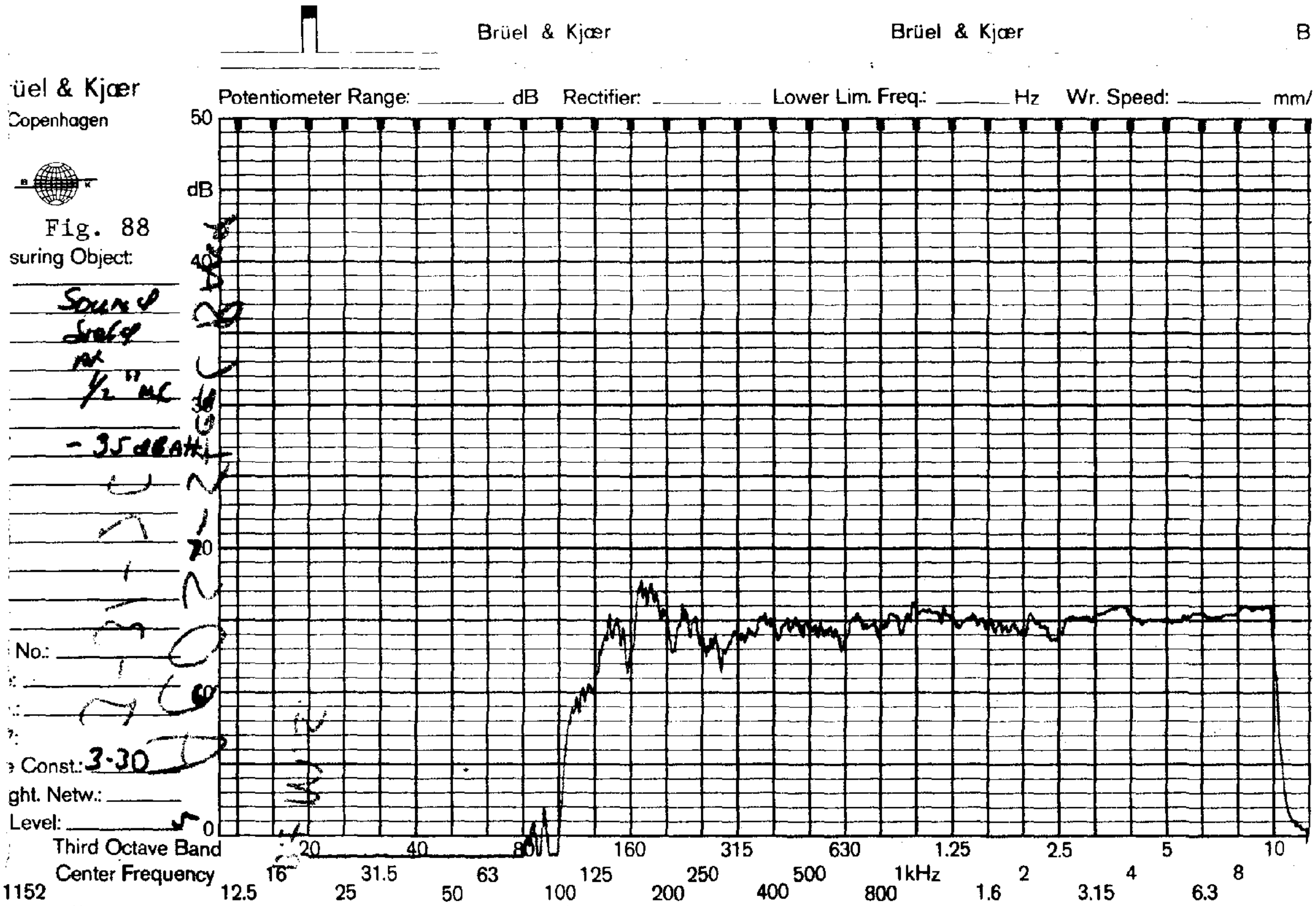
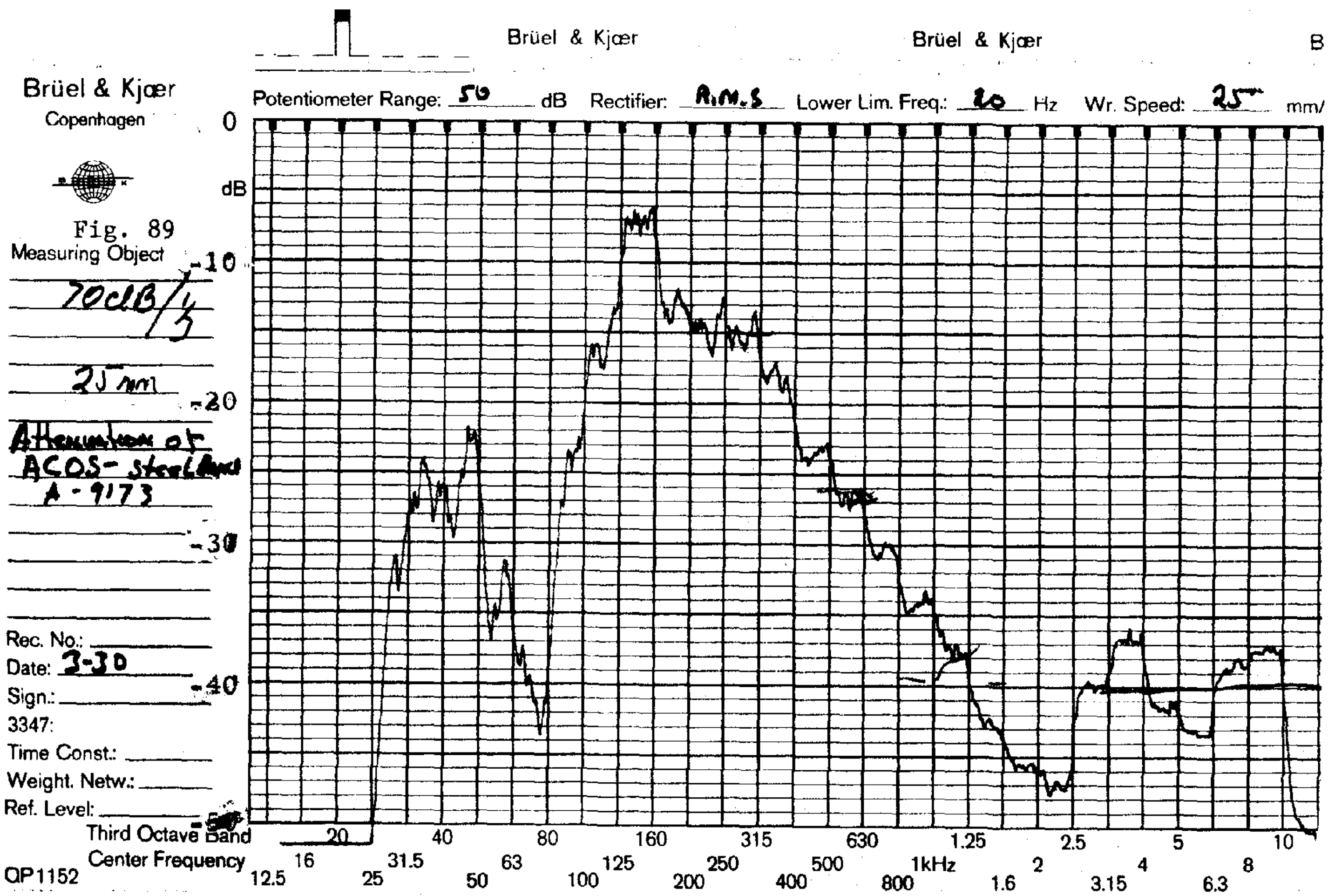


Fig. 87 PERFORMANCE OF ACOS A9000 ELECTRONIC HEARING PROTECTOR



Brüel & Kjær
Brüel & Kjær
Potentiometer Range: dB Rectifier: Lower Lim. Freq.: Hz Wr. Speed: mm/
Copenhagen
Measuring Object:
Sound Field at 1/2" MC
-35 dB Att.
No.:
Const.: 3-30
Weight. Netw.:
Level:
Third Octave Band
Center Frequency
50
dB
40
30
20
10
0
12.5 16 20 25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 800 1kHz 1.25 1.6 2 2.5 3.15 4 5 6.3 8 10
1152

Fig. 88



Brüel & Kjær
Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB Rectifier: R.M.S Lower Lim. Freq.: 20 Hz Wr. Speed: 25 mm/
Measuring Object
70dB/1/3
25mm
Attenuation of ACOS-Steel Band A-9173
Rec. No.:
Date: 3-30
Sign.:
3347:
Time Const.:
Weight. Netw.:
Ref. Level:
Third Octave Band
Center Frequency
0
dB
-10
-20
-30
-40
-50
12.5 16 20 25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 800 1kHz 1.25 1.6 2 2.5 3.15 4 5 6.3 8 10
QP1152

Fig. 89

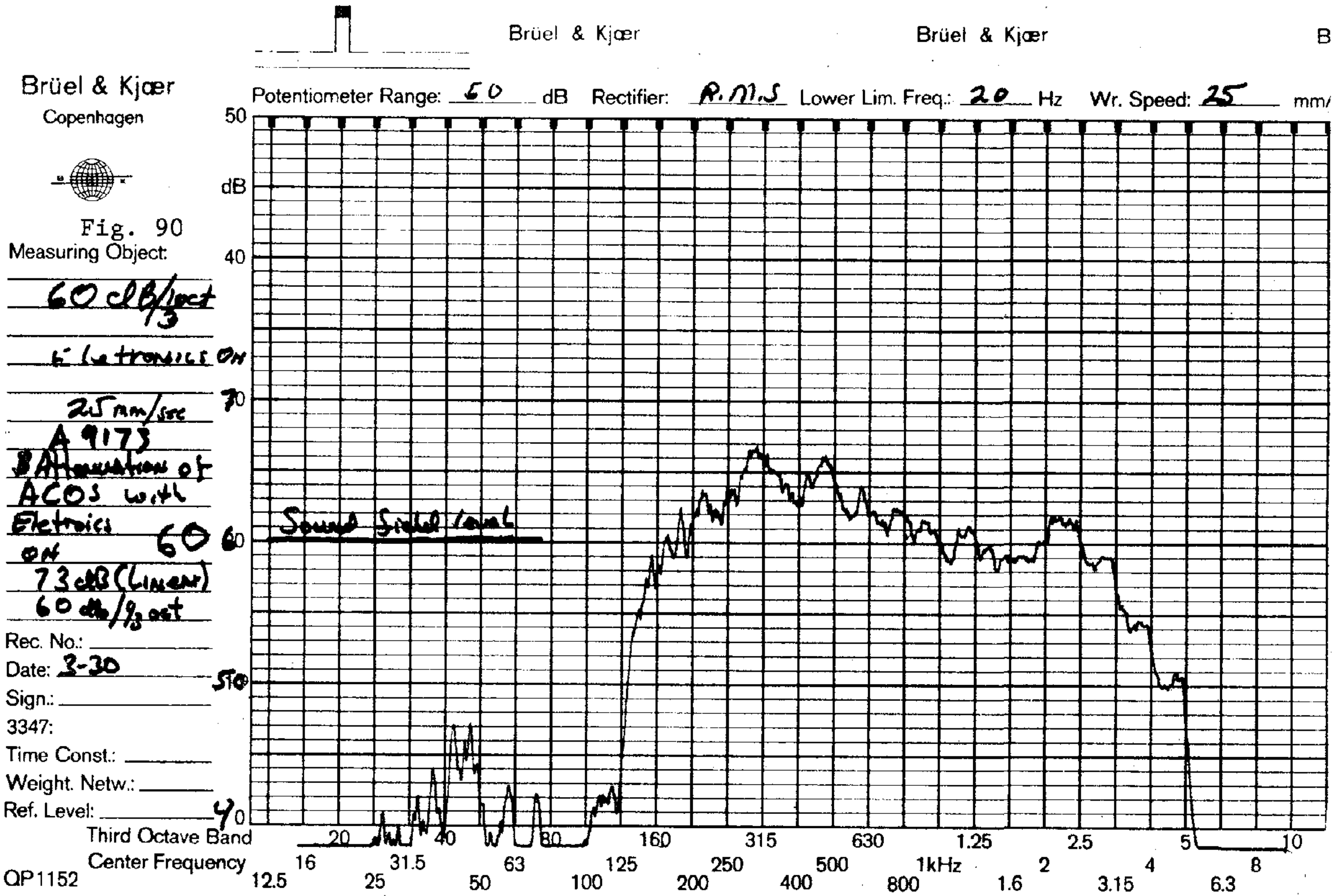

Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB
Rectifier: R.M.S
Lower Lim. Freq.: 20 Hz
Wr. Speed: 25 mm/
Measuring Object:
60 dB/1/3 oct
Electronics ON
25 mm/sec
A 9173
Attenuation of ACOS with Eletroics ON
60
73 dB (Linear)
60 db/1/3 oct
Rec. No.:
Date: 3-30
Sign.:
3347:
Time Const.:
Weight. Netw.:
Ref. Level:
Sound field level
50
dB
40
70
60
50
40
Third Octave Band
Center Frequency
12.5
16
20
25
31.5
40
50
63
80
100
125
160
200
250
315
400
500
630
800
1kHz
1.25
1.6
2
2.5
3.15
4
5
6.3
8
10
QP1152

Fig. 90

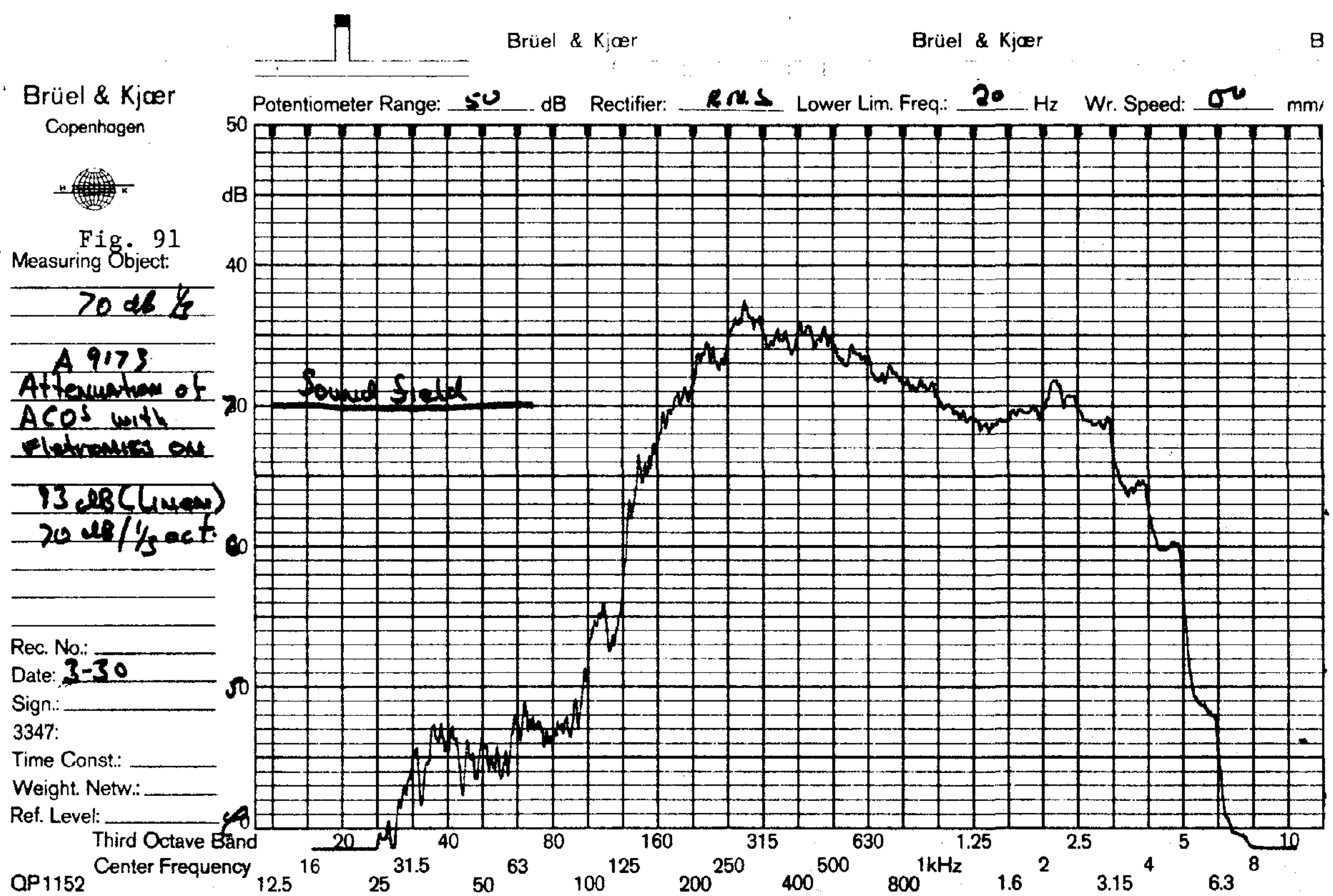

Brüel & Kjær
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB
Rectifier: RMS
Lower Lim. Freq.: 20 Hz
Wr. Speed:
Measuring Object:
70 db 1/3
A 9173
Attenuation of ACOS with Electronics ON
73 dB (Linear)
70 dB/1/3 oct.
Rec. No.:
Date: 3-30
Sign.:
3347:
Time Const.:
Weight. Netw.:
Ref. Level:
Sound field
50
dB
40
70
60
50
40
Third Octave Band
Center Frequency
12.5
16
20
25
31.5
40
50
63
80
100
125
160
200
250
315
400
500
630
800
1kHz
1.25
1.6
2
2.5
3.15
4
5
6.3
8
10
QP1152

Fig. 91

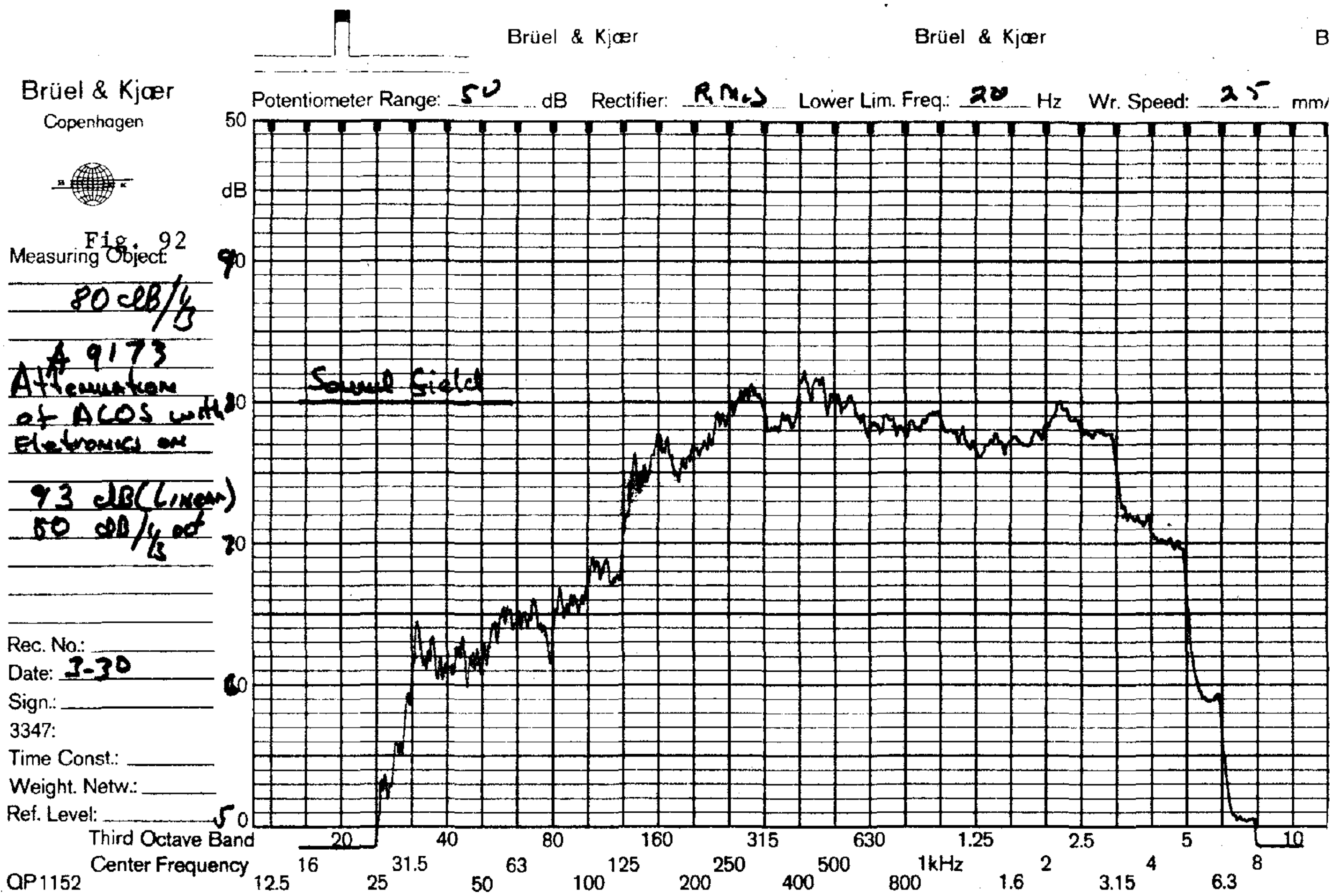

Brüel & Kjær
Brüel & Kjær
B
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB
Rectifier: R.M.S
Lower Lim. Freq.: 20 Hz
Wr. Speed: 25 mm/
dB
Measuring Object:
80 dB/1/3
A 9173
Attenuation of ACOS with Electronics on
93 dB (Linear)
80 dB/1/3 oct
Sound field
Rec. No.:
Date: 3-30
Sign.:
3347:
Time Const.:
Weight. Netw.:
Ref. Level: 50
Third Octave Band
Center Frequency
QP1152
12.5 16 20 25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 800 1kHz 1.25 1.6 2 2.5 3.15 4 5 6.3 8 10

Fig. 92

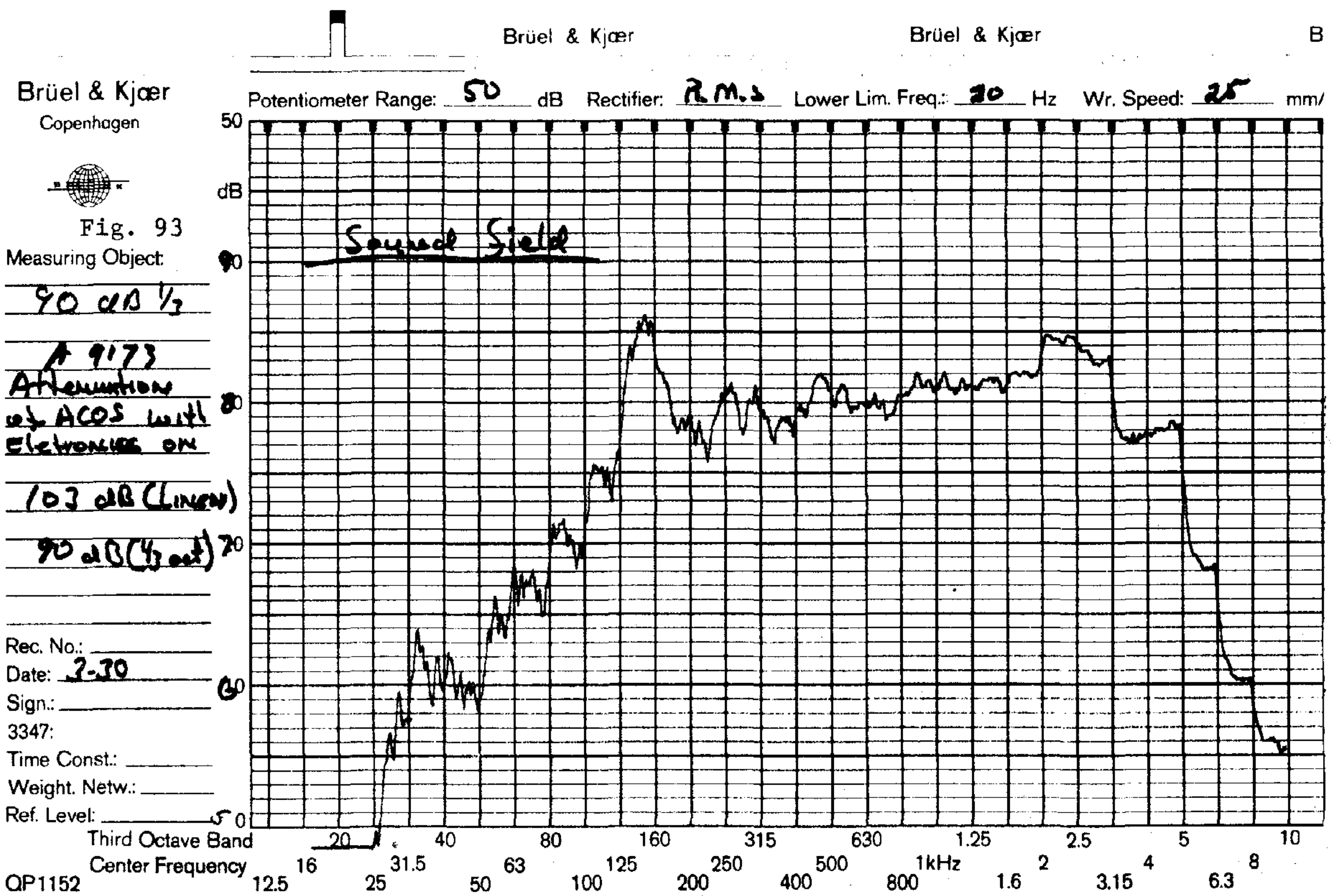
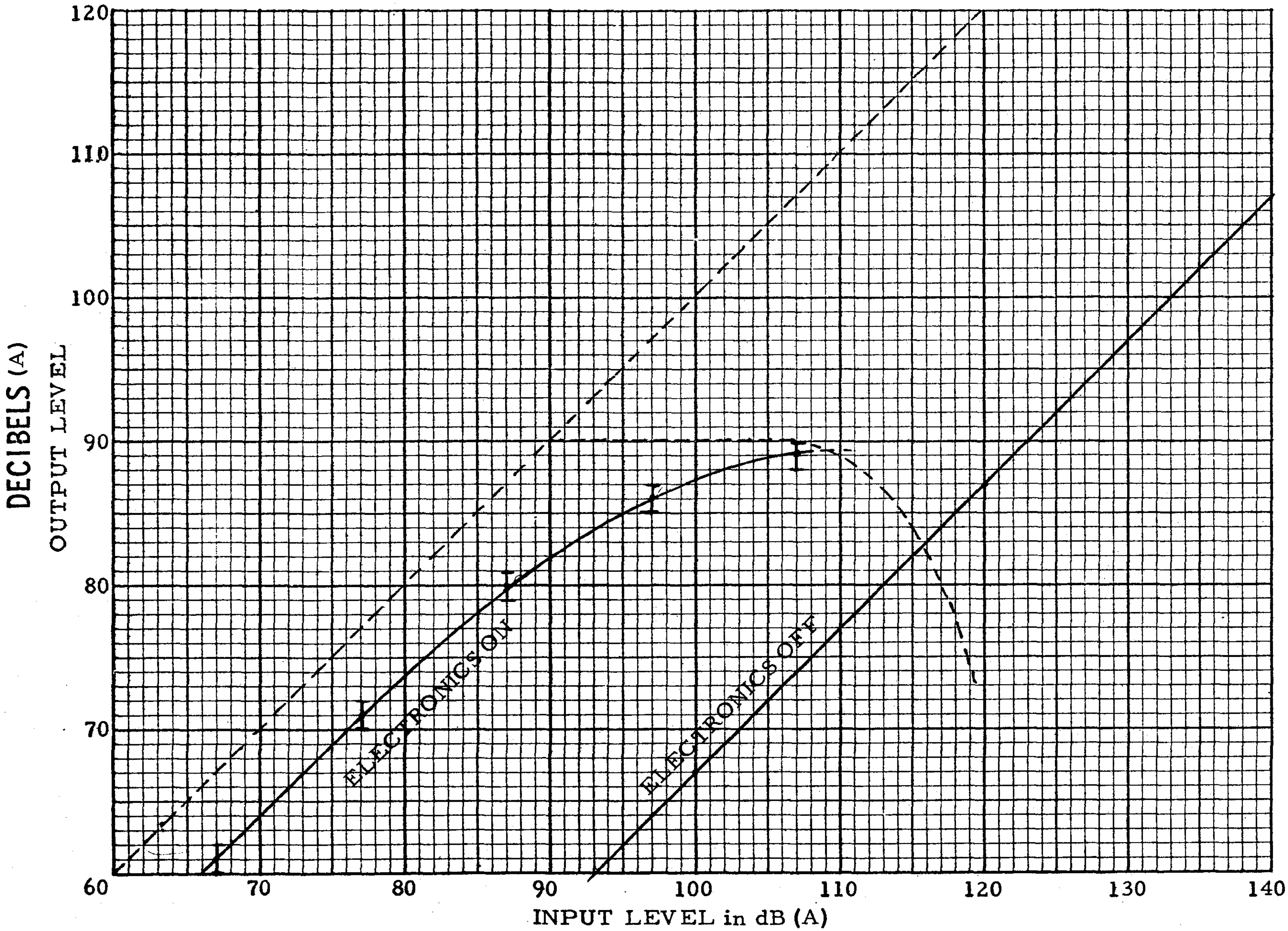

Brüel & Kjær
Brüel & Kjær
B
Brüel & Kjær
Copenhagen
Potentiometer Range: 50 dB
Rectifier: R.M.S
Lower Lim. Freq.: 20 Hz
Wr. Speed: 25 mm/
dB
Sound field
Measuring Object:
90 dB 1/3
A 9173
Attenuation of ACOS with Electronics on
103 dB (Linear)
90 dB (1/3 oct)
Rec. No.:
Date: 3-30
Sign.:
3347:
Time Const.:
Weight. Netw.:
Ref. Level: 50
Third Octave Band
Center Frequency
QP1152
12.5 16 20 25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 800 1kHz 1.25 1.6 2 2.5 3.15 4 5 6.3 8 10

Fig. 93

FIGURE 94



ELECTRONIC TRANSFER FUNCTION FOR ACOS A9173
PEAK LIMITING EAR DEFENDER

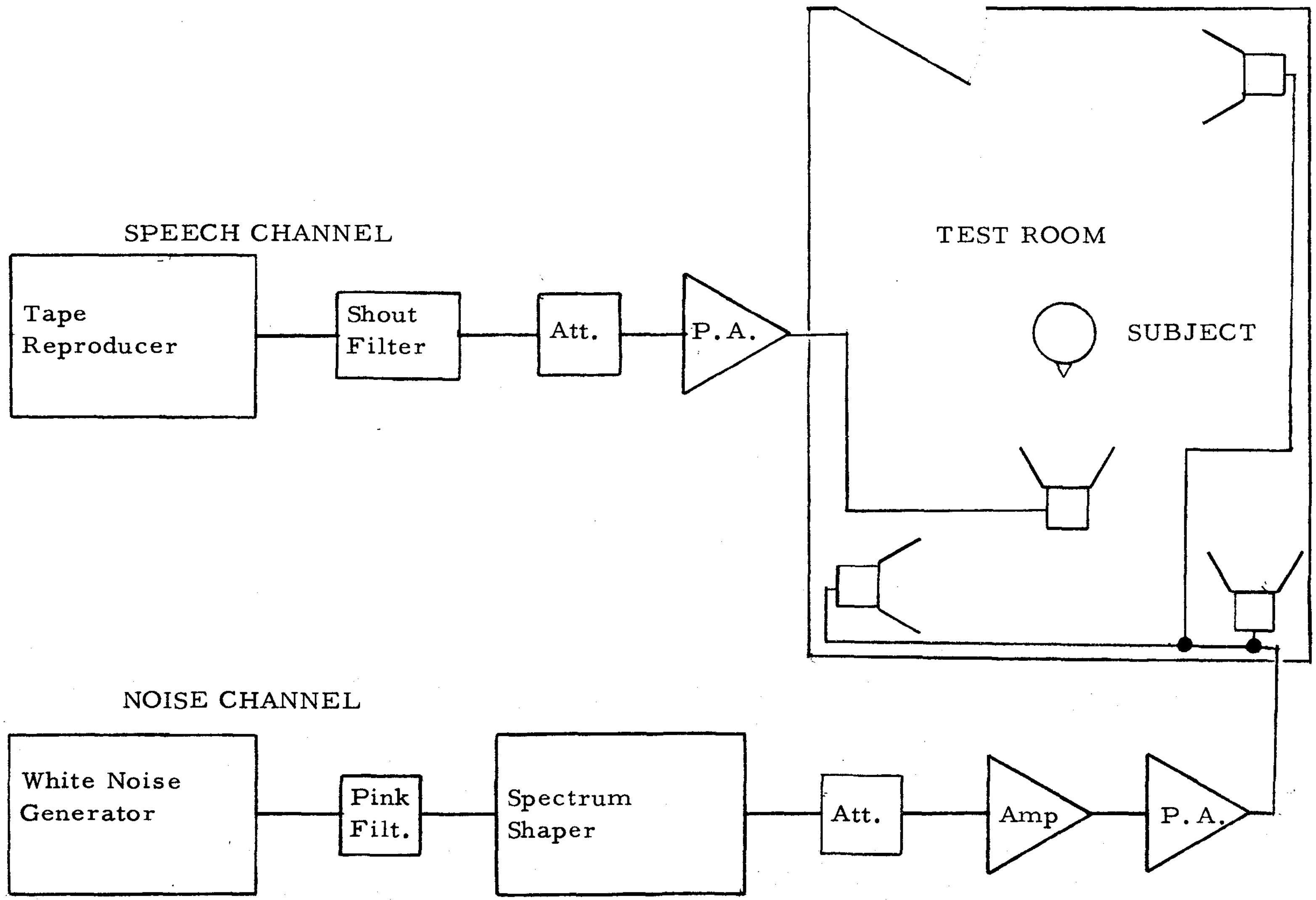


Fig. 95 Block diagram of test facility for evaluation of hearing protectors.

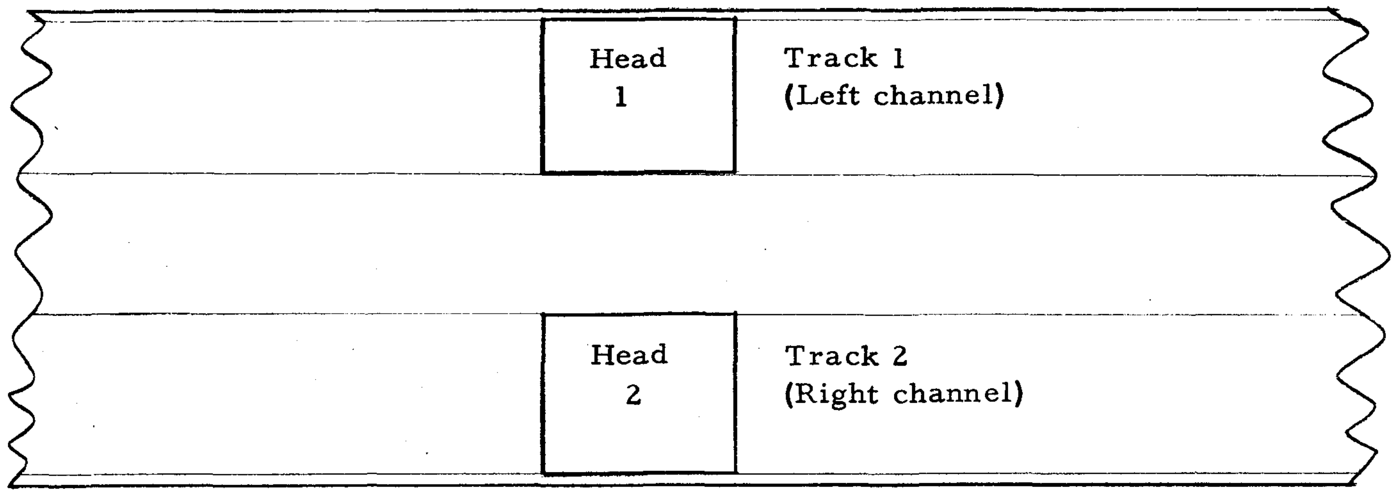


Fig. 96 Two-Track Stereo head configuration for 1/4" tape.

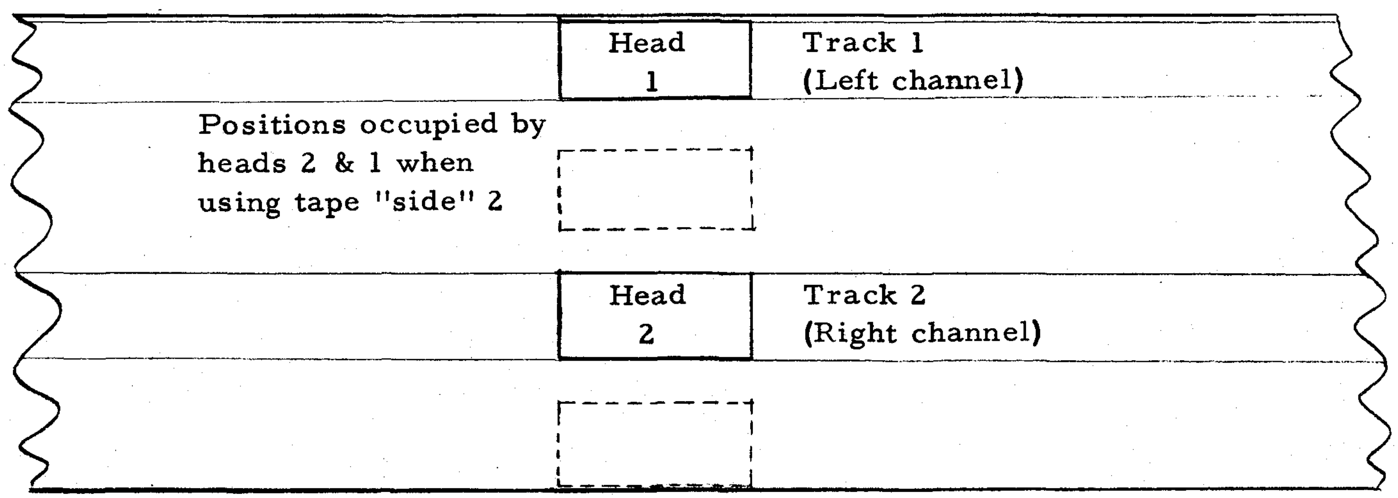


Fig. 97 Four-Track Stereo head configuration for 1/4" tape.

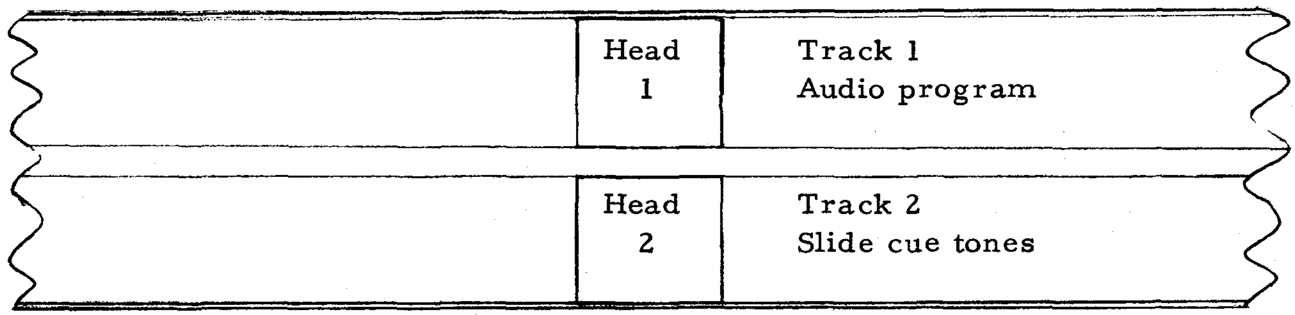


Fig.98 Two-channel configuration for 3M cassette system.

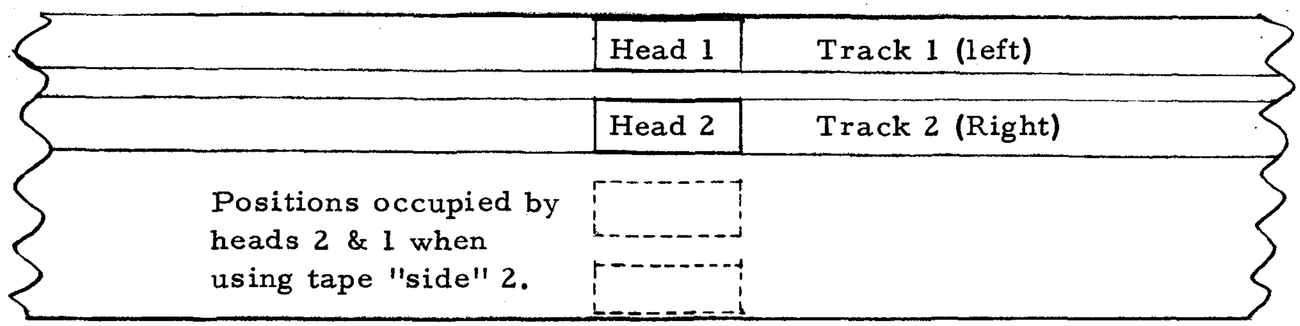


Fig.99 Stereo track configuration for cassette tapes.

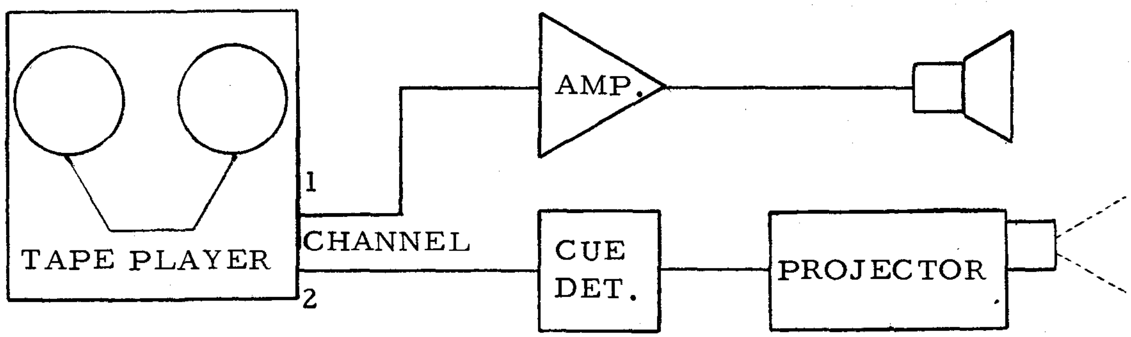


Fig.100Equipment connections for automated slide presentation.

APPENDIX A

7 Mining Machinery Noise Spectrum

To simplify tests of speech intelligibility in underground coal mine communications systems, it is desirable to use a masking noise that is typical of that produced by mining machinery. Although each machine makes its own peculiar sound, the task of deriving a "typical" spectrum is not insurmountable. The 1/3 octave spectra published by Derzay & Goodwin* provides a useful set of measurements for this purpose.

From the above report, the octave band spectra of mining machinery commonly found in underground coal mines was selected. The maximum levels reported for four major machines are plotted in Fig. A1. The minimum levels measured for the continuous miner (undercutter) are also plotted to show the lower boundary for the noise spectra. Fig. A1 shows that the spectra of each of these machines exhibits a general downward slope as frequency increases above 125 Hz. It should also be noted that the roof bolter octave band levels decrease rapidly below 125 Hz.

It is obvious that all machines do not operate all the time. Thus the noise exposure from any given machine depends partly on its sound level and partly on the proportion of time it is actually operating. These factors have been taken into account in the calculation of an "exposure factor" based on "questioning the worker or company official or from knowledge of the job". This "exposure factor" is the "estimated exposure received by workers exposed to the noise source divided by the ACGIH permissible exposure". Thus the exposure factor can be used to weight the individual machine spectra according to the proportion of noise exposure produced by each machine.

To calculate the weighted average maximum spectrum, the octave band maximum levels are multiplied by the individual exposure factors from Table A1. The weighted levels in each octave band are then added and divided by the sum of the exposure factors. The resulting octave band levels are plotted in Fig. A2 as the Weighted Average Maximum noise spectrum.

The dominance of the roof bolter spectrum is evident in the low frequency slope below 125 Hz. This slope is approximately 9 dB per octave. The high frequency slope above 125 Hz is approximately -5 dB per octave. An RC network has been constructed which matches the smoothed high frequency slope characteristics. No attempt has been made to duplicate the low frequency slope because it is peculiar to the roof bolter and may not be representative of general mining machinery.

* Evaluation of Noise in the Mineral Industry, Internal Report, U. S. Bureau of Mines Mineral Industry Health Program, December 19, 1969.

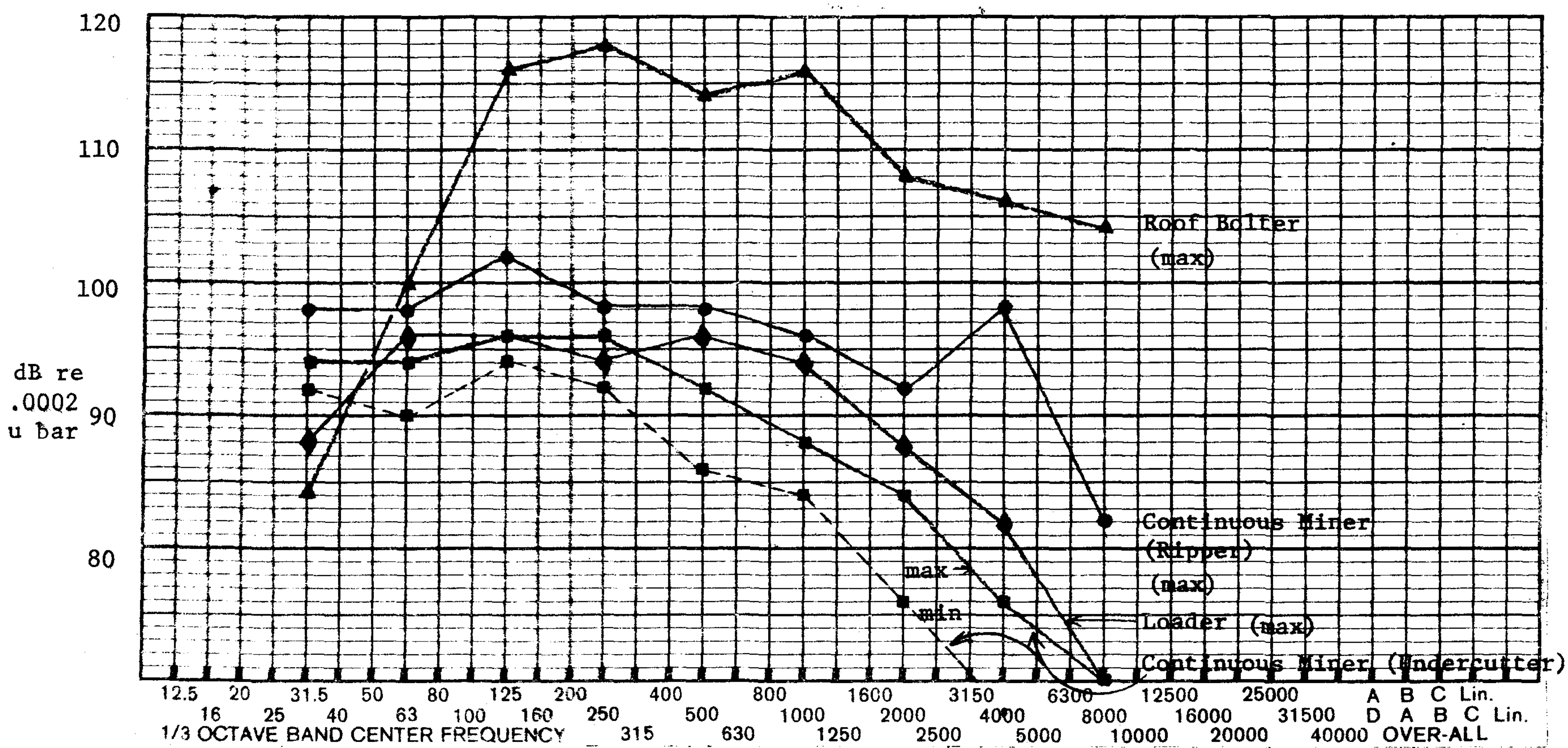


Fig. A1 Spectra of Underground Coal Mining Machines.

TABLE A1

Calculation of Weighted Average Maximum Lp from Octave Band Spectra

	Octave Bands								
lachine	31.5	63	125	250	500	1000	2000	4000	8000
:ontinuous Miner (Ripper)	98	98	102	98	98	96	92	98	82
:F 1.11	108.8	108.8	113.2	108.8	108.8	106.6	102.1	108.8	91.0
:ontinuous Miner (Undercutter)	94	94	96	96	92	88	84	76	70
:F .67	63	63	64.3	64.3	61.6	59.0	56.3	50.9	46.9
Roof Bolter	84	100	116	118	114	116	108	106	104
:F = 2.47	207.5	247.0	286.5	291.5	281.6	286.5	266.8	261.82	256.9
Loader	88	96	96	94	96	94	88	82	70
EF = .91	80.1	87.4	87.4	85.5	87.4	85.5	80.1	74.6	63.7
ΣdB	459.4	506.2	551.4	550.1	539.4	537.6	505.3	496.1	458.5
WAM	89.0	98.1	106.9	106.6	104.5	104.2	97.9	96.1	88.9

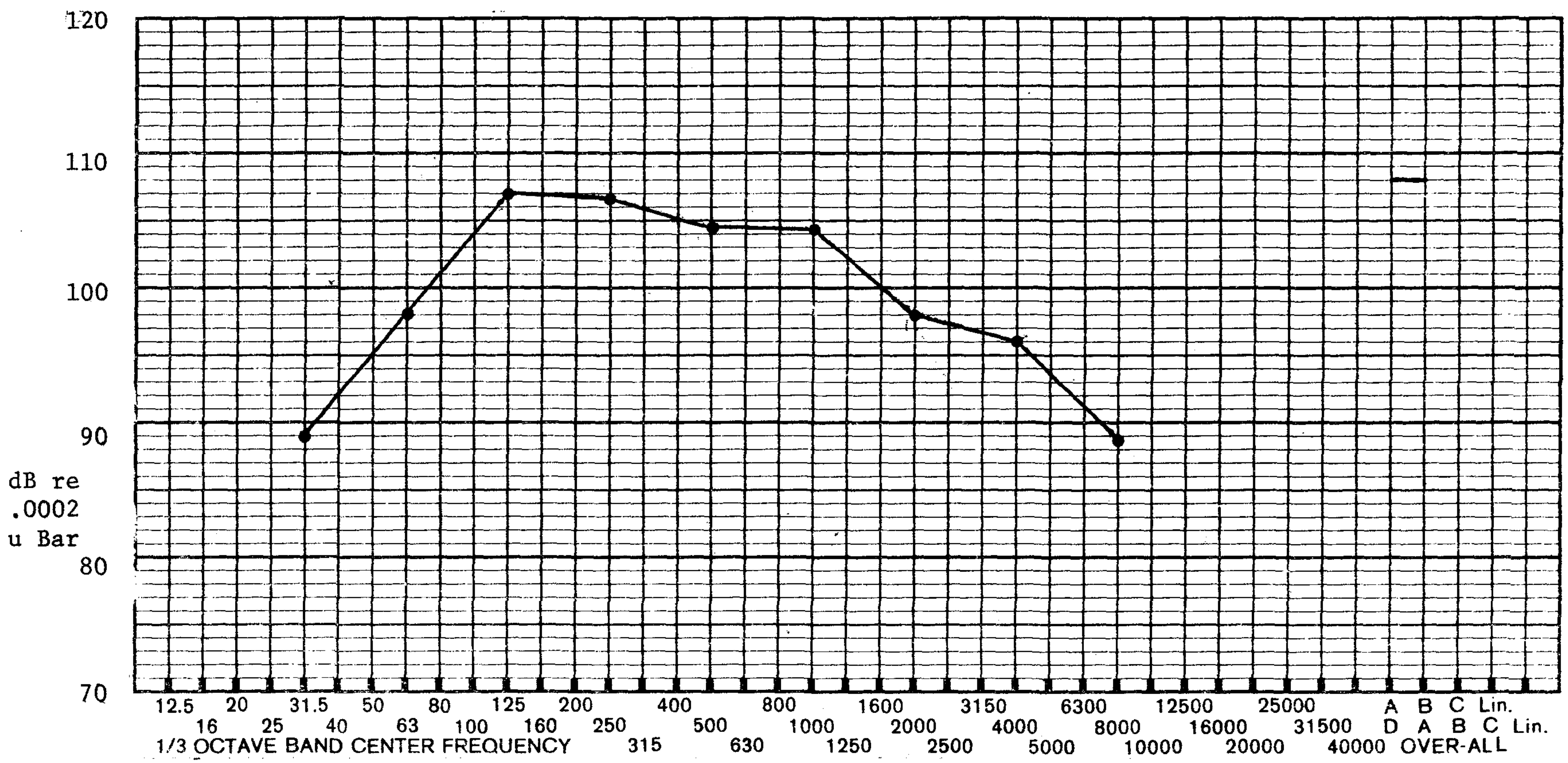


Fig. A2 - Weighted Average Maximim Noise Spectrum for Underground Coal Mines

Appendix B

8. Recruitment

The presence of non-linearities in the perception of loudness by patients with noise-induced hearing loss was first noted by Haberman (1890) in his now classic paper on "Boilermaker's disease". The phenomenon was named "recruitment" by E. P. Fowler (1928) and was defined as an abnormally rapid increase in loudness (sound magnitudes as perceived by a listener) when the sound level is increased. Stated otherwise, a given increase in sound level created a greater increment in the sensation of loudness for the recruiting ear than for the non-recruiting ear. A person with recruitment experiences extreme annoyance for loud sounds and a decreased range of sound levels that are confortable for listening. Speech sounds are distorted and sound "foggy" or "blurred". Fowler (1963) charaterized the most common difficulty of the recruiting listener: "The result is he will say, 'Do not shout at me,' and yet if you lower your voice, even a trifle, he will say, 'Don't mumble,' or 'I can't hear your voice, it seems loud enough, but I cannot understand what you are saying.'"

Having defined the phenomenon of loudness recruitment, Fowler went on to observe that it was absent in cases of middle ear pathology and concluded that the phenomenon arose from some neural malfunction of the hearing mechanism (1928, 1936, 1937, 1938). In 1948, Dix, Hallpike, and Hood were able to refine Fowler's conclusion by demonstrating that the recruitment phenomenon was limited to cases of cochlear and organ pathology. This finding has been supported by Luscher (1950), Eby and Williams (1951), and Dix (1965). Indeed, Harris (1953) noted that those pathologies showing loudness recruitment involve some mechanical damage within the cochlea as contrasted with a strictly neural dysfunction. Mygind (1950) shares this view suggesting that recruitment is indicative of a "conductive impairment" within the cochlea. Thus, it is generally agreed that the phenomenom of loudness recruitment is a pathological manifestation caused by structural injury to the cochlea.

When recruitment is present subsequent to noise exposure the affects are usually noticed first at high frequencies. As indicated above, the range of acceptable sound level is very narrow. Speech or warning signals occurring at higher or lower levels will be unintelligible. A communication system with automatic level control (as proposed for the electronic hearing protectors) may not necessarily regulate the sound at the proper level for a recruiting ear. Because the sound level is regulated electronically, asking the person talking to repeat the message at a lower level may not change the message level at the recruiting ear. Since recruitment is generally noticed first at high frequencies, the problems can be minimized for most individuals by preventing troublesome high frequencies from reaching the ear. Since 3000 Hz is an adequate upper frequency limit for good intelligibility and naturalness (Fletcher 1953) sound above this frequency can be reduced. As a practical consideration, a high frequency roll-off of at least 12 dB/octave should provide significant attenuation of high frequency energy above 3000 Hz to minimize most problems associated with recruitment.

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