THE DEVELOPMENT AND DESIGN OF AN AUDIO AMPLIFIER SYSTEM CONTROLLED BY ACOUSTICAL BACKGROUND NOISE

A Thesis

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ABSTRACT

This thesis contains a report on a method of automatically ensuring a suitable level of background music in premises having a varying background noise level. In order to ascertain the technical and economic suitability of this method, a completely transistorized prototype was designed, developed, constructed, and tested.

PREFACE

This thesis has been the result of much experimental trial and error that was conducted in an attempt to obtain a commercially and economically realizable variable gain amplifier to control background music level in the presence of a varying background noise. The method that was examined was that of sensing and measuring the level of only a small portion of the frequency spectrum of the background noise. In order that the music signal would not interfere with this measurement, this same portion of the frequency spectrum was eliminated from the music signal by means of a filter. The sensed noise signal in turn was used to control a variable gain amplifier which in turn controlled the music level, thus maintaining what was considered to be the desired relationship between the music signal and the background noise.

The thesis has been presented in a manner that shows the different attempts made to find a solution to this problem. In addition, a detailed description has been given of a practical device that was designed and constructed in order to ascertain its economic and technical feasibility.

The work involved in this thesis was essentially a practical engineering project. In order to arrive at design specifications, judgements were made that were based on limited statistical evidence. This was found necessary due to the lack of information available in references and also due to the cost and time restrictions that had to be observed in the empirical and subjective determination of design criteria.

The author wishes to acknowledge the help of Mr. R. M. Heald for his assistance in developing the broad objectives of the project and the aid of all others who kindly volunteered their time and considered opinions for the subjective experiments required to obtain the objective. Appreciation is also expressed to Professor J. P. C. McMath for his helpful suggestions.

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CHAPTER ONE

INTRODUCTION TO THE PROBLEM

1-1 STATEMENT OF THE PROBLEM

In the past few years, background music has been used more and more in hotels, restaurants, chain stores, etc.

In discussions with a supplier* of this type of music, the following information was obtained:

Background music is used for various reasons, ranging from offering a relaxing atmosphere in restaurants, to increasing the efficiency of workers in office buildings and factories. This music is not the highest in quality. Instead it is purposely recorded with a rather narrow dynamic range. Also when played back, it is further restricted in its dynamic range by a compressor type of amplifier. The reason for this is that the music is normally played at approximately the same level as that of the room background noise and thus it must be relatively constant in level if it is not to be masked by the noise. As well, the music is restricted in its upper frequency range to less than 7 K.C. In practice, this limited range is found to give a satisfactory quality. It is important

*Rimac Ltd., Winnipeg, Manitoba, Canada.

that the music not have a harsh character, however, and thus amplitude distortion is kept to a minimum.

If this music is to have the desired effect, it must be loud enough to be heard over the background noise. That is, the always prevalent background noise must not mask the music signal so that only an occasional bar or two of music is heard. The listener should not be straining to hear the melody of the music. On the other hand the music should not be so loud that it is annoying and thus be considered as an additional noise.

In practice, it has been found that the program level must be adjusted accurately with respect to the background noise to obtain the desired result. Unfortunately, the background noise level is, in many cases, a large variable. Restaurants, for instance, have a background noise level that varies greatly, depending on the number of people being served.

If the music level is adjusted during a quiet period, it is masked by the noise when the noisy period is reached. On the other hand, if the music level is adjusted during the noisy period, it becomes annoyingly loud when the quiet period is reached. In both cases, an undesirable situation is the result.

In order to overcome this problem, an amplifier which can automatically maintain a correct music level in the presence of the variable background noise is needed. This, basically, is the problem.

The foregoing has applied to music. In some locations, commercials are inserted between musical selections. Again it is important that these commercials are received at the correct listening level with respect to the background noise. Thus the same problem occurs for these spoken commercials as for the music.

1-2 PREVIOUS METHODS AT SOLVING THE PROBLEM AND THEIR SHORTCOMINGS

There have been past attempts at solving this problem. The two main methods that have been used are:

(1) The background noise level is sensed only during the silent interval between musical selections. This sensed noise in turn controls a variable gain amplifier or mechanical switch which adjusts the music level for the next selection. Altec Lansing Corp., Anahein, California, has used this principle of operation in manufacturing a commercial unit described as the "NOALA Control Panel, Model No. 7464."

Problems encountered with this method are:

(i) A rather complicated arrangement is required to sense the silent interval and thence switch in the noise measuring circuitry.

(ii) If the background noise level increases or decreases during a musical selection, the music to noise ratio will be unsatisfactory until the next silent interval is reached at which time a readjustment can take place.

(iii) In some cases the silent interval can be very short, and thus if the background noise level happens to be momentarily low or high during this brief noise sensing period, an unsatisfactory music level will be produced.

(2) The other method is to form a bridge with one arm being controlled by the incoming music level and another arm by the background noise level. If the noise level goes up or down, it unbalances the bridge. The output of the bridge controls a variable gain amplifier through which the music signal passes. The music level is automatically changed in such a way as to restore the bridge balance and thus maintain a satisfactory music signal to noise ratio. This method of operation has been described by Kannenberg[15].

Problems encountered with this method are:

(i) There is no positive way of separating the acoustic music signal from the room noise. Thus, an increase in the acoustic music signal appears as noise to the noise sensing microphone and this calls for more music level. Thus, this causes a positive feedback signal that can ultimately cause the variable gain amplifier to operate at maximum gain. This problem can partially be overcome by operating the music signal at a low level and by physically separating and orienting the speakers associated with the music signal away from the noise sensing devices. Nevertheless, this constitutes a serious problem.

(ii) During the silent intervals between musical selections, the bridge becomes unbalanced, calling for greater gain in the variable gain amplifier. Upon the music starting to play, the high gain of the variable gain amplifier produces an unsatisfactorily loud music signal until the bridge is rebalanced. In some designs, this problem is partially overcome by means of circuitry that senses the cessation of the music signal, and then prevents the bridge from affecting the variable gain amplifier until the next musical selection commences.

1-3 PROPOSED METHOD OF SOLVING PROBLEM

The method that was investigated for the purpose of this thesis was to sense the level of only a portion of the frequency spectrum of the background noise. This part of the spectrum was eliminated from the music signal so that only the noise signal was measured. This noise signal in turn controlled a variable gain amplifier which, by controlling the music level, maintained a suitable music to noise ratio.

This method was considered to have eliminated the objections of previous methods since:

(1) This method of operation continuously sampled the background noise and thus was always in control of the music level.

(2) This method of operation separately measured the background noise without interference from the music signal. This prevented possible positive feedback between the music signal and the noise measuring channel.

The restriction on using this method was that it could be satisfactory only if the noise spectrum remained the same regardless of the loudness of the noise, or the type of the noise. The reason for this was that since only a small portion of the noise spectrum was used for measuring the whole spectrum of noise, it was necessary that this small portion of noise have always the same relative level with respect to the whole noise spectrum in order to be considered representative of the background noise. This could only be satisfied if the noise spectrum did not change as the noise level changed.

If this method were restricted to use in premises where the prevalent noise was due to the conversation of men and women, then this assumption was considered valid. If the prevalent background noise was machine noise or other types of noise that had a varying noise spectrum, then this method was not considered completely valid. The intended application for this method was in restaurants, hotels, office buildings, etc., where the noise spectrum was considered to be relatively constant, and was made up mostly of talkers. Thus it was expected that the only variable would be that of the loudness of the noise.

Since the actual noise spectrum was of no concern for the intended application, it was decided that measurements of noise spectra at different restaurants and stores were not required and thus none were taken.

In the search for available references, this method of operation was not found to be described in the literature. Thus, references given do not cover all aspects of the development and testing that were undertaken. Much of the development work consisted of limited subjective measurements. As a result, the discussions and conclusions of this thesis are not and cannot be considered rigorous.

CHAPTER TWO

DEFINING OF DESIGN OBJECTIVES

The method of approach to be taken having been decided upon, it was found necessary to determine the design objectives that would subsequently be used in the design and construction of a prototype unit. In part, these objectives were arrived at by employing empirical methods and by considering the accepted practice of the industry.

The initial objectives determined at this point were:

(1) the range of variation in gain that the prototype device would be required to have in order to give a satisfactory music signal to noise ratio over the normal range of background noise levels found.

(2) the music signal to noise ratio that was preferable for different background noise levels within the normal range encountered.

(3) the maximum music signal to noise ratio that would be necessary to give satisfactory results.

(4) the maximum harmonic distortion that the prototype device could contribute to the music signal.

2-1 BACKGROUND NOISE RANGE

In order to determine the normal range of background noise levels found in restaurants, stores, etc., noise level measurements were made in various premises with the following results:

TABLE 1

BACKGROUND NOISE RANGES IN VARIOUS ESTABLISHMENTS

Establishment	Minimum Noise Reading (db above ref.)	Maximum Noise Reading (db above ref.)
Embassy Grill Restaurant	60	75
T. Eaton Co. Canada Ltd. (main floor)	57	75
The Bay (main floor)	52	72
Loblaws Portage & Ainslie	57	73
Safeway Portage & Ainslie	55	70

These and all other acoustic noise readings were made with a Bruel and Kjoer sound level meter, type 2203. This device was calibrated in db with reference to 0.0002 microbar sound pressure. In this thesis, all acoustic readings and measurements were with reference to the 0.0002 microbar reference level. In general, due to the fluctuations of the reading, the experimental error of all acoustic measurements was considered to be ±2 db. In some cases, this error was reduced by averaging a large number of readings.

The readings in Table I were obtained as follows:

(1) Embassy Grill Restaurant - To determine the lowest prevalent noise level, measurements were taken at a time when the restaurant was almost empty. There were perhaps one or two customers, but no talking was taking place. The tables were clear and so there were no noises of dishes being moved. The high reading measurements were taken at approximately 10:00 A.M. and 3:00 P.M. At these times the restaurant was completely filled, and incoming and outgoing traffic was at a maximum. These measurements were taken at four different times and the maximum and minimum readings are given in Table I.

(2) Eatons, Bay, Loblaws, Safeway - To determine the lowest noise levels, measurements were taken just after the times of opening. It was considered that measurements taken at these times represented the lowest noise levels that would normally occur. At Eatons and the Bay, the measurements were made on the main floors at various locations. At Loblaws and Safeway, the measurements were made in the quietest areas, generally the area where the

canned goods were stocked. The measurements for the high readings were taken at what was considered the busiest times. For all locations this was considered Friday night and Saturday afternoon. The measurements were made just once at each location and perhaps did not represent the maximum levels that might be encountered. However, the stores were quite busy, and it was decided that these levels were sufficiently indicative.

After examining these measurements, it was decided that the variable gain amplifier should have a range sufficient to control the music level with background noise levels varying from 50 to 80 db above reference. This range was greater than the Table I measurements indicated as being necessary. However, it was considered that these measurements, although indicative, were not completely representative of all stores and restaurants. Therefore, it was decided to expand the range of control in an attempt to cover as many applications as possible.

2-2 DESIRABLE MUSIC SIGNAL TO BACKGROUND NOISE RATIO

No references could be found which stated the preferable ratio of background music to the prevailing background noise over the normal range of background noise levels found in hotels, stores, etc. Thus an experiment was conducted to determine the desirable music signal to noise ratio. Tests were made at various background noise levels ranging from 50 to 80 db above reference. This range was indicated by the previous results.

For a noise source, the background noise of the Embassy Grill restaurant was recorded on tape. This noise was representative of the busy interval. It was considered that this noise was reasonably typical of other restaurants and stores as it consisted mainly of noises generated by both men and women talking, moving their chairs, walking, etc. This noise was then played back through a speaker. Music was also played back through another speaker placed immediately below the noise speaker. The music signal was of a type meant for background music purposes. That is, the dynamic range was severely restricted and the upper frequency range was limited to less than 7 K.C. The amplifier and speaker that were used were not of the highest quality but were typical of the type normally used for background reproduction. Thus they represented a normal arrangement.

The room in which the testing took place was considered to be acoustically "live" and to be approximately equivalent acoustically to a typical restaurant or store. It was considered that this arrangement was suitable for testing purposes insofar as being economically possible.

In order to obtain consistent results with reasonable economy, it was decided to use expert male personnel of ages between 32 and 50, who had been experienced in adjusting music levels in customers' premises. These people were asked to listen to both the music and the background noise, and to adjust the noise level for what they thought the proper music signal should be in the presence of the background noise. The testing was conducted by the author who was also one of the listeners. Thus, a certain interaction of opinion as to what was the correct level may have been present during the tests. However, this was not considered a serious defect.

Both the music level and the background noise level were separately measured with the Bruel and Kjoer sound level meter. This was repeated for a range of noise levels from 50 to 80 db above reference with the following results:

TABLE 2

DESIRABLE MUSIC SIGNAL TO NOISE RATIO

FOR VARIOUS NOISE LEVELS

Music Level (db above ref.)	Noise Level (db above ref.) Listener A	B	C	Avg.	Avg. S/N
50	51	52	50	51.0	-1.0
60	60	61	61	60.8	-.8
70	71	73	72	72.0	-2.0
80	81	83	81	81.7	-1.7

It seemed unusual that the desirable music to noise ratio should be negative even for a small and select number of people. It was thought that the explanation might have been that the noise was impulsive in nature, having "peaks" and "valleys" in its level that varied with time. The background music, on the other hand, was purposely limited in its dynamic range and thus produced a more constant level that tended to fill in the "valleys" of noise. Thus the background music may actually have been heard most of the time even with a negative signal to noise ratio, and thus gave a satisfactory result. No further work was done concerning this unusual result as it was considered that the results obtained were satisfactory.

It also seemed unusual that there was such a close agreement between the opinions of the listeners. It was thought that the reason for this may have been as follows:

Normally, the human ear is considered to be a poor absolute measuring device. However, in comparing differences of level of two signals, the human ear is considered to be relatively sensitive and accurate[2]. In this case, the difference in level existed between the constant background noise and the constant music signal. Also, the tests were conducted using persons having experience in adjusting the background music level in various establishments. These persons would tend to have similar opinions as to the preferable music signal to noise ratio.

From these limited tests, it was decided that the music to noise ratio should be considered as being independent of the actual background noise level between 50 and 80 db above reference and that the desirable signal to noise should be approximately equal to minus one db.

Next, two of the listeners were asked to adjust the music level to a point which would be considered too loud and also to a point considered too soft. This was done at a background noise level of 72 db above reference only since the previous results had shown good consistency.

TABLE 3

ACCEPTABLE RANGE OF MUSIC SIGNAL TO NOISE RATIO

	Music Level of Listener (db above ref.) in the Presence of 72 db of Noise A	B	Avg.
Too Soft	68	69	68.5
Too Loud	76	76	76

From these tests, it was considered that the desirable music to noise ratio should be -1 db with a tolerance of ±4 db between "too soft" and "too loud." However, as a design objective, it was decided that the music signal to noise ratio should be kept to within ±2 db of that considered as desirable and thus be well within the range considered as being acceptable.

2-3 STATEMENT OF DESIGN OBJECTIVES FOR THE PROTOTYPE DEVICE

In order to proceed with the development, the following design objectives for the prototype device were specified on the basis of the previous tests combined with engineering judgement:

(1) The range of control of the gain of the variable gain amplifier was to be 30 db. There was to be a one to one relationship between the input

noise signal and the gain of the variable gain amplifier. The maximum deviation from the one to one relationship was to be ± 2 db; i.e., for any x db increase or decrease in noise, the variable gain device should increase or decrease in gain x db ± 2 db. These requirements were arrived at from the results of Tables 2 and 3.

(2) The desirable music signal to noise ratio was decided as being -1 db. However, it was considered that the music level might be played louder than this in some establishments. Also, it had been standard practice in the industry to play voice commercials at 7 db higher than the music. This would mean the desirable <u>voice signal</u> to noise ratio would be 6 db. Besides this, it was decided to allow a further 6 db for misadjustments, personal taste, etc., so that the normal maximum expected signal to noise ratio would be 12 db. It was decided that with a 12 db signal to noise ratio, there must be negligible acoustic positive feedback between the music signal and the sensed noise signal. That is, the effect of the music signal on the sensed noise signal should be such as to cause less than a 2 db change in the music level.

(3) The distortion of the variable gain amplifier should be less than 2% at peak music levels. This was considered as meeting the accepted practice of the industry.

(4) The band of noise used for noise sampling should be as representative as possible of the whole spectrum of noise.

CHAPTER THREE

DESIGNS, TRIALS, PROBLEMS AND THEIR SOLUTIONS

The main problem encountered at this point was to determine how wide the noise sampling bandpass should be and in what part of the noise spectrum it should be located. Too wide a bandpass would tend to adversely affect the quality of music but a very narrow bandpass would excessively increase the cost of the filters used. No information could be found describing the effect of slots in the music. Thus, it was decided to undertake experimental trials in order to arrive at a solution. The number of trials had to be limited due to time and cost restrictions. In order to arrive at a reasonable result, three trials were undertaken. The first trial considered using a relatively low frequency band (120 to 190 cps) for sensing the noise level. The second trial considered using the band of frequencies above 6 K.C. for sensing the noise level. The third and final trial considered using a mid-frequency band (between 1,000 and 2,000 cps) for sensing the noise level.

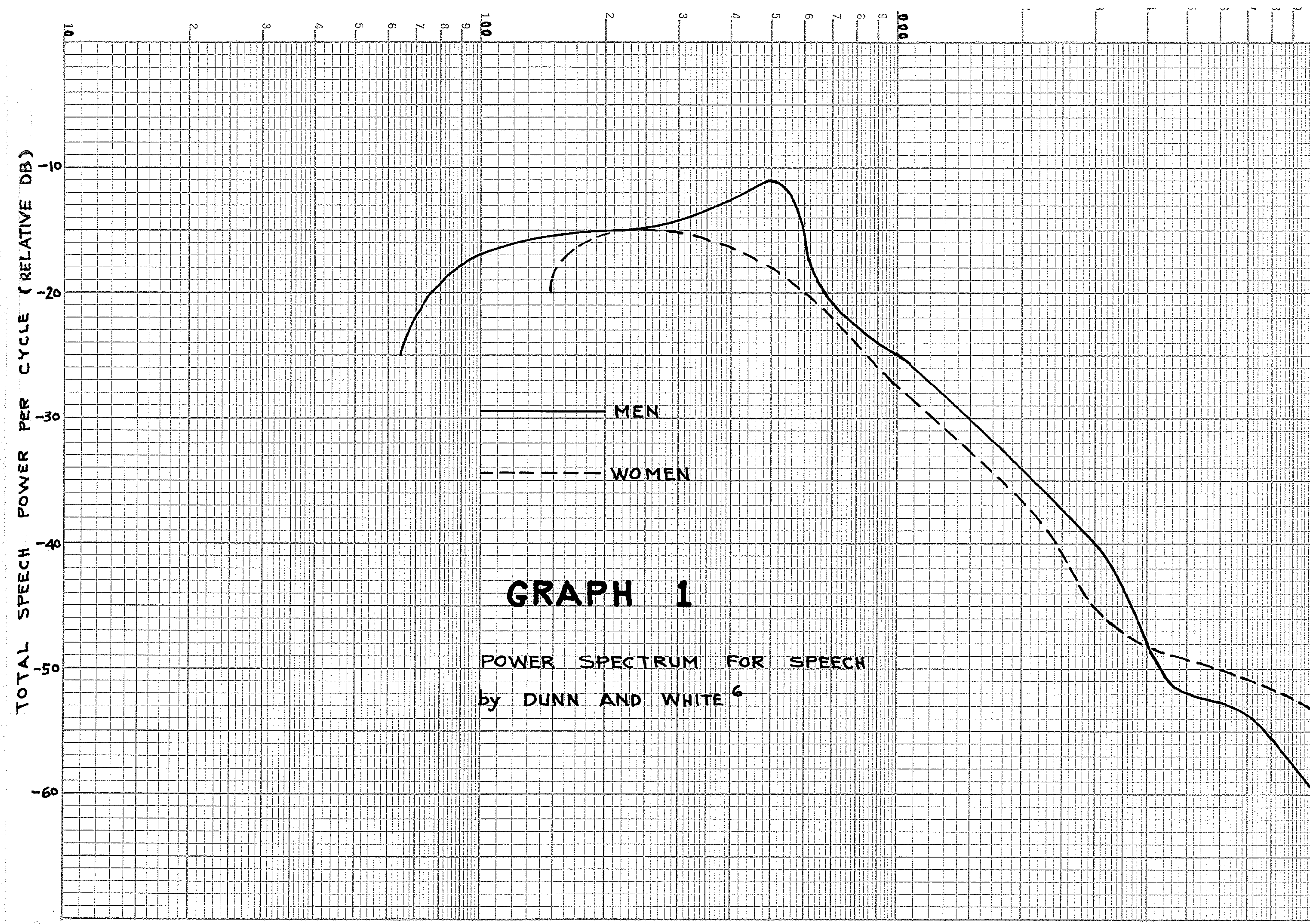
Upon the completion of these tests, a prototype unit was designed and constructed. At all times during the design and construction, economy, as well as performance was considered.

3-1 LOW FREQUENCY TRIAL

It was first considered logical that the low frequency end of the spectrum (120 to 190 cps) would be the best portion of the spectrum to use for sampling the noise. Reference GRAPH 1, as given by Dunn and White[6], it can be seen that most of the power in speech is located below 500 cps, and thus a 120 to 190 cps noise amplifier would not require a high gain amplifier. Further, listening tests conducted by the author tended to show that a slot of 120 to 190 cps in the music signal did not materially change the quality of the music signal for background music purposes.

Accordingly, a trial arrangement was made as shown in FIG. 1. The response of the slot rejection filter that was placed in the music channel and that of the bandpass filter that was placed in the noise channel are shown in GRAPH 3. The electrical circuit of each filter is shown in FIG. 2.

These filters were designed by the so called "cut and try" method. This was considered the most suitable method since:



GRAPH 1
POWER SPECTRUM FOR SPEECH
by DUNN AND WHITE [6]
MEN
WOMEN
TOTAL SPEECH POWER PER CYCLE (RELATIVE DB)
-10
-20
-30
-40
-50
-60

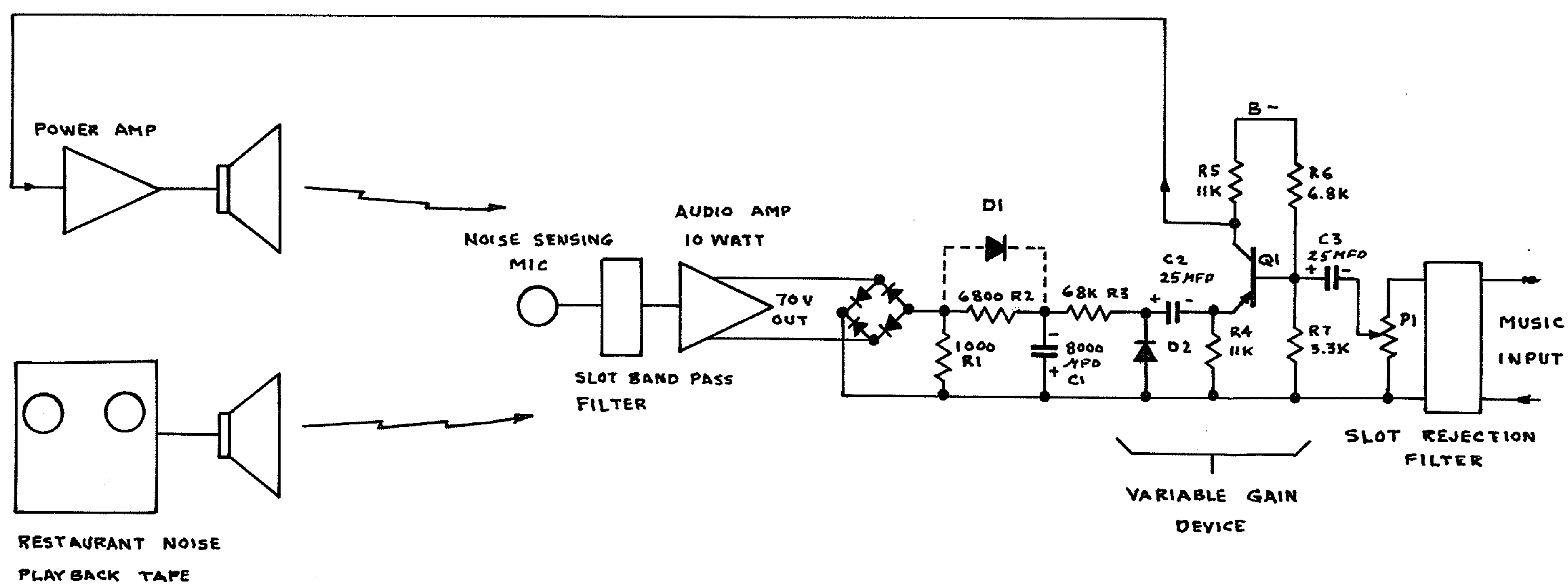


FIG. 1

PROTOTYPE ARRANGEMENT FOR CONSTANT MUSIC SIGNAL TO NOISE RATIO WITH VARYING NOISE LEVEL

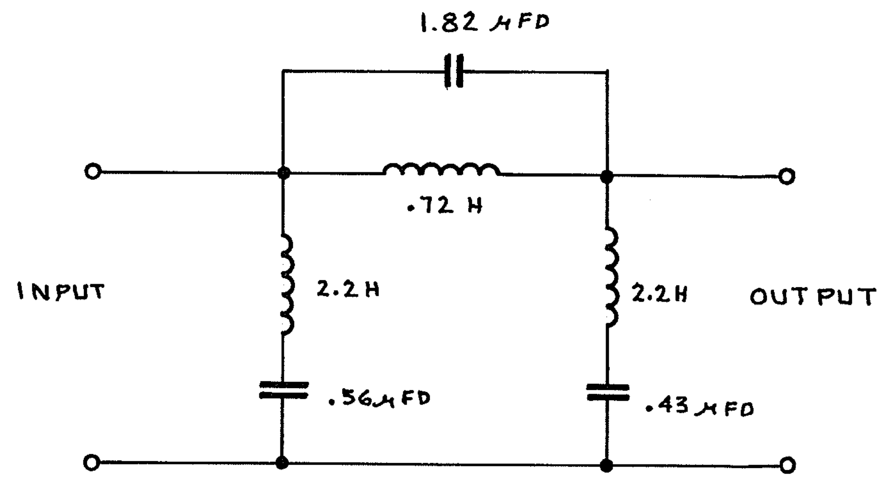


FIG. 2A

LOW FREQUENCY (120 TO 190 CPS) SLOT
REJECTION FILTER

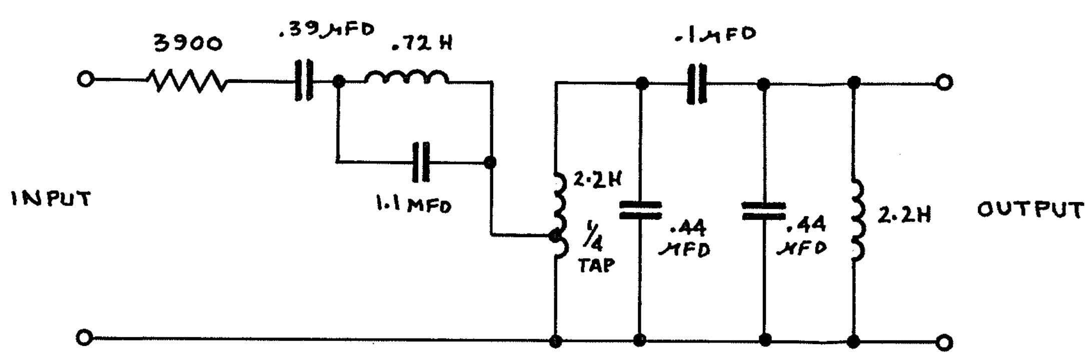


FIG. 2B

LOW FREQUENCY (120 TO 190 CPS) SLOT
BAND PASS FILTER

(1) Only a limited number of inductor values were readily available and these had relatively low quality factors ranging from 10 to 30 at 150 cps.

(2) It was expected that this method would result in obtaining the required filters more quickly.

The general formulae and forms of the filters were obtained from available designers handbooks[10, 12]. Thence a few trial calculations were undertaken to familiarize the author with the relative values required. After selecting available inductors and capacitors to form a rough prototype, the capacitor of each arm of the filter was adjusted in discrete steps to resonate with the inductor of the arm at the desired frequency. The sink and source impedances were then experimentally determined for the required response.

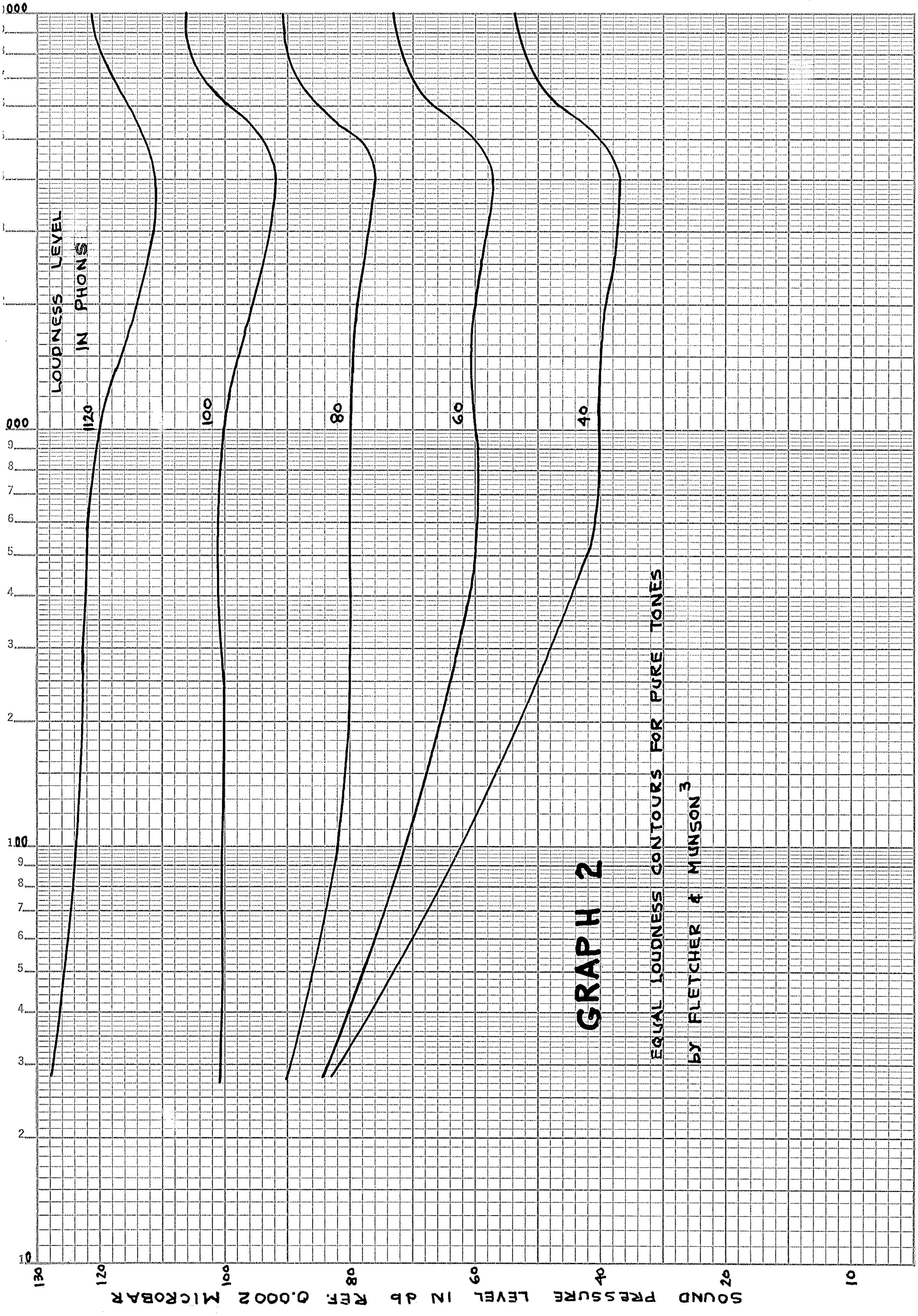
After a few trials, the author obtained a practical knowledge of the relative values required to obtain the required filter characteristics.

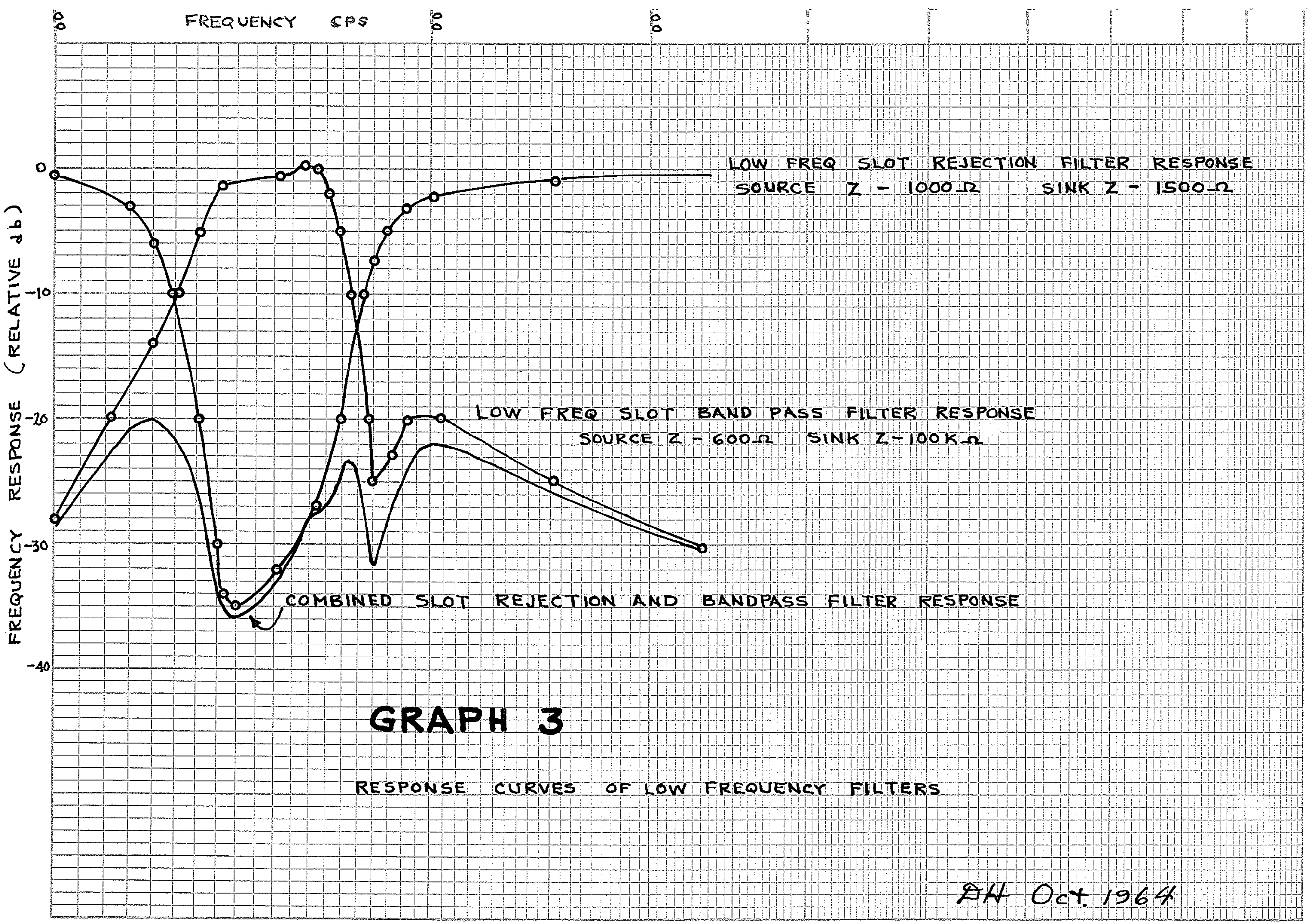
In the adjustment of the filters, it was found to be important that the frequency characteristic of the slot rejection filter match that of the bandpass filter in order to obtain minimum interaction between the music signal and the noise sensing channel. In some cases, this required a slight readjustment of the capacitors of the filters to obtain the best result.

In order to evaluate this system, the taped background noise from the Embassy Grill restaurant was used as the noise signal and the music signal was sent through the variable gain device. It was found that this system worked well. For instance, if the background noise level was increased 10 db, then the music signal level increased 10 db and the desired music to noise ratio was maintained.

However, it was soon evident that other noises, besides speech, were causing annoying changes in the music level. Trucks passing by, or the shuffling of feet were found to have an excessive effect on the level of the music. This can be seen in the Fletcher-Munson[3] equal loudness curves reproduced in GRAPH 2; i.e., in the 120 to 190 cps band, low frequency noise, due to trucks, etc., does not affect the ear appreciably. However, since the noise sensing device was controlled by this band of noise, it was sensitive to low frequency noises of trucks, etc., and resulted in an annoyingly large increase in music level where there should only have been a small change in level. For this reason, the 120 to 190 cps slot was not considered satisfactory as the noise sensing spectrum.

An additional problem had presented itself at this time. As shown in FIG. 1 the filter for the rectified noise spectrum consisted of a simple resistor-capacitor


GRAPH 2
EQUAL LOUDNESS CONTOURS FOR PURE TONES
BY FLETCHER & MUNSON 3
LOUDNESS LEVEL IN PHONS
120
100
80
60
40
SOUND PRESSURE LEVEL IN db REF. 0.0002 MICROBAR
130
120
100
80
60
40
20
10



GRAPH 3

RESPONSE CURVES OF LOW FREQUENCY FILTERS

DH Oct. 1964

(R2, C1) filter with a long time constant. In order to overcome large fluctuations in the noise signal, it was decided, after numerous listening tests, that the time constant should be approximately one minute. However it was considered that when the background noise dropped, that the gain of the variable gain amplifier did not drop fast enough, thus giving a too loud music level for a period of time. In order to overcome this, a diode D1 as shown dotted in FIG. 1 was added. This gave a much faster decay time in relation to the rise time of the smoothing filter. As can be seen, the diode D1 only conducted when capacitor C1 was discharging. Thus the discharge time constant was R1, C1 and the charge time constant was (R2 + R1)C1. The ratio of charge time to discharge time was

$$\frac{(R2 + R1)C1}{R1\ C1}$$

$$= \frac{R2 + R1}{R1}$$

which was equal to in this case to:

$$= \frac{6800 + 1000}{1000}$$

$$= 7.8 \text{ to } 1.$$

This arrangement was considered to help greatly in giving a good follow of the variable gain device to background noise variations.

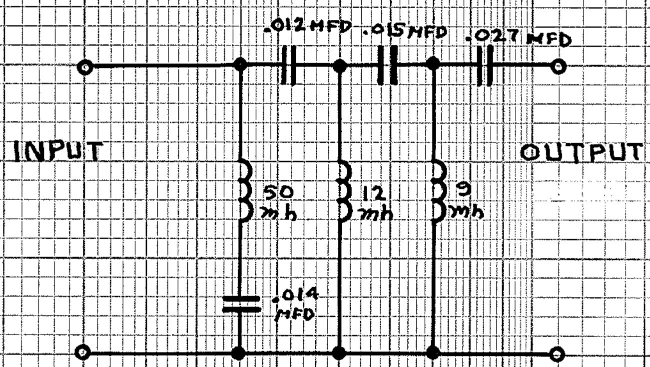
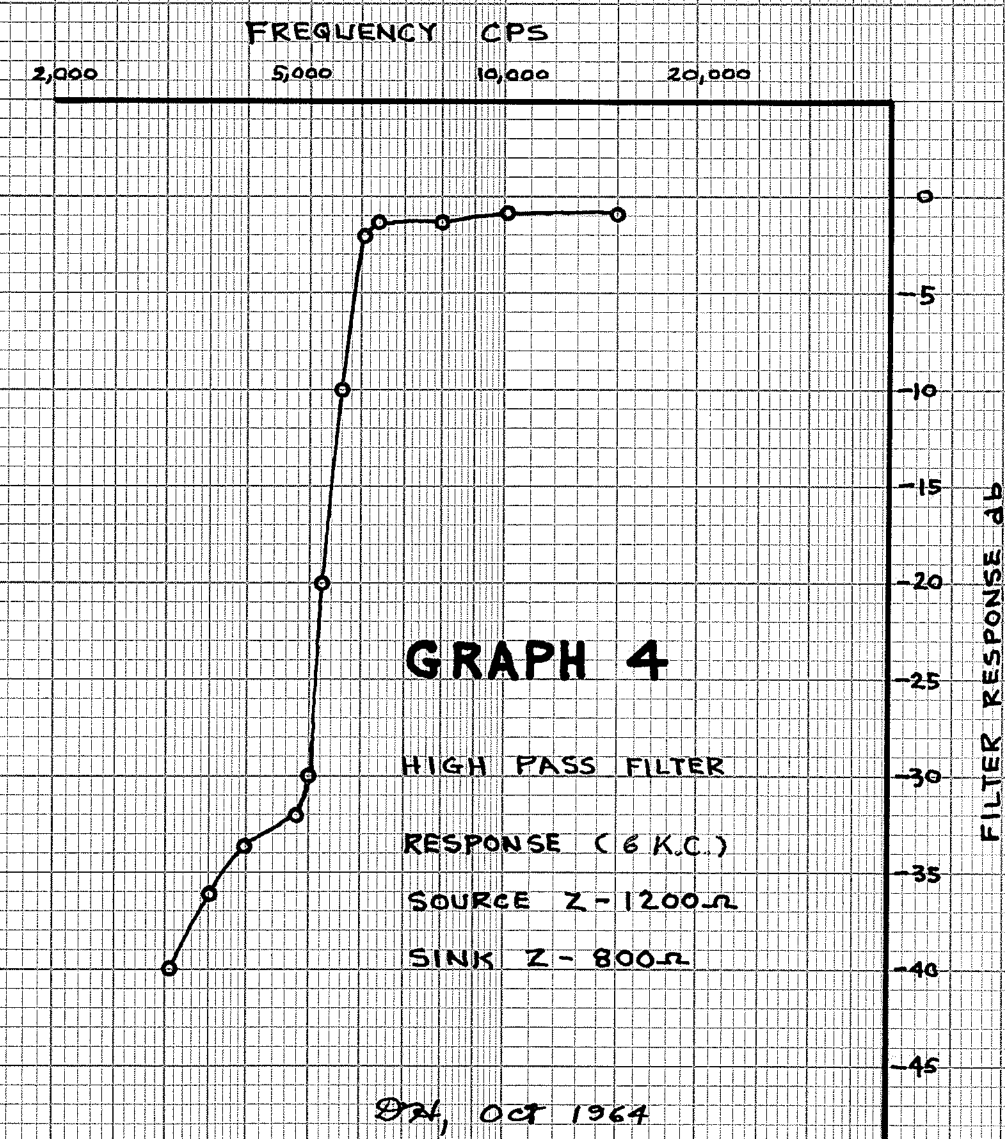


FIG. 3

HIGH PASS FILTER (6 K.C.)



GRAPH 4

HIGH PASS FILTER

RESPONSE (6 K.C.)

SOURCE Z - 1200Ω

SINK Z - 800Ω

DH, Oct 1964

3-2 HIGH FREQUENCY TRIAL

It was next considered logical to attempt to use the background noise that existed above 6 K.C. as the noise sensing spectrum. This choice was made for the following reason.

Normally the background music signal is purposely restricted to below 7 K.C. or less. In practice, this limited range has been found to give a satisfactory quality. It was thought that restricting the music upper frequency range to 6 K.C. would cause little degradation in quality. Thus, since the music signal could be limited to frequencies less than 6 K.C., the noise sensing channel could use all of the noise spectrum above 6 K.C. for the purpose of sensing the background noise level.

Accordingly, a trial was conducted in the same manner as shown in FIG. 1 except a 6 K.C. high pass filter was used in place of the slot bandpass filter.

As with the previous filters, this filter was constructed with available inductors on a "cut and try" basis. The circuit is shown in FIG. 3 and the frequency response is shown in GRAPH 4.

The available music signal tape playback system was found to have negligible music signal above 6 K.C. and so no low pass filter was required to suppress the music above 6 K.C.

This trial was considered unsuccessful for the following reasons:

(1) The noise sensing microphone was directional above 6 K.C. and thus could not sense noise from all directions equally well. A more omnidirectional microphone could have been obtained but it was considered that its cost would be excessive for this practical application.

(2) The amount of noise in this band was very limited and was produced mostly by the "s" consonant. The resulting noise was found to have a large ratio of peak noise signal to average signal. This made it more difficult to smooth out the average noise and thus control the variable gain amplifier.

(3) This noise was found to be very low in level and thus a high gain, low noise amplifier with associated high costs would be required.

3-3 MID FREQUENCY TRIAL

It was next considered logical to sense the background noise in the middle of the noise spectrum. This required the insertion of a slot rejection filter in the middle of the music spectrum in order to keep the music signal out of the noise sensing channel. It was at first thought that there would be references available that would aid in the determination of the width and frequency band of the slot for the least interference with the musical signal. A diligent search was made but unfortunately no information on this matter could be found. Thus, it was decided to determine a suitable slot by means of listening tests. This is believed to be an original attempt at assessing the effect of slots on the quality of the music. The first choice was a slot between 1,150 to 1,650 cps. This spectrum was chosen for the following reasons:

(1) As given by Dunn and White[6], and shown on GRAPH 1, page 21, the band of frequencies between 1,150 and 1,650 cps had the same general shape for both men and women talkers. At some frequencies of the noise power spectrum, this was not so. For instance, in the vicinity of 500 cps, men talkers had a pronounced peak in their voice energy whereas women talkers did not. Thus a narrow noise sensing band in the 500 cps vicinity would have tended to respond to men talkers more readily than to female talkers even if they both produced the same effective noise loudness. Naturally, this would not have been desirable.

(2) This slot was above the highest fundamental of most musical instruments[8]. Therefore, it was thought that removing this band of frequencies from the musical spectrum would not be as likely to cause deteriation to

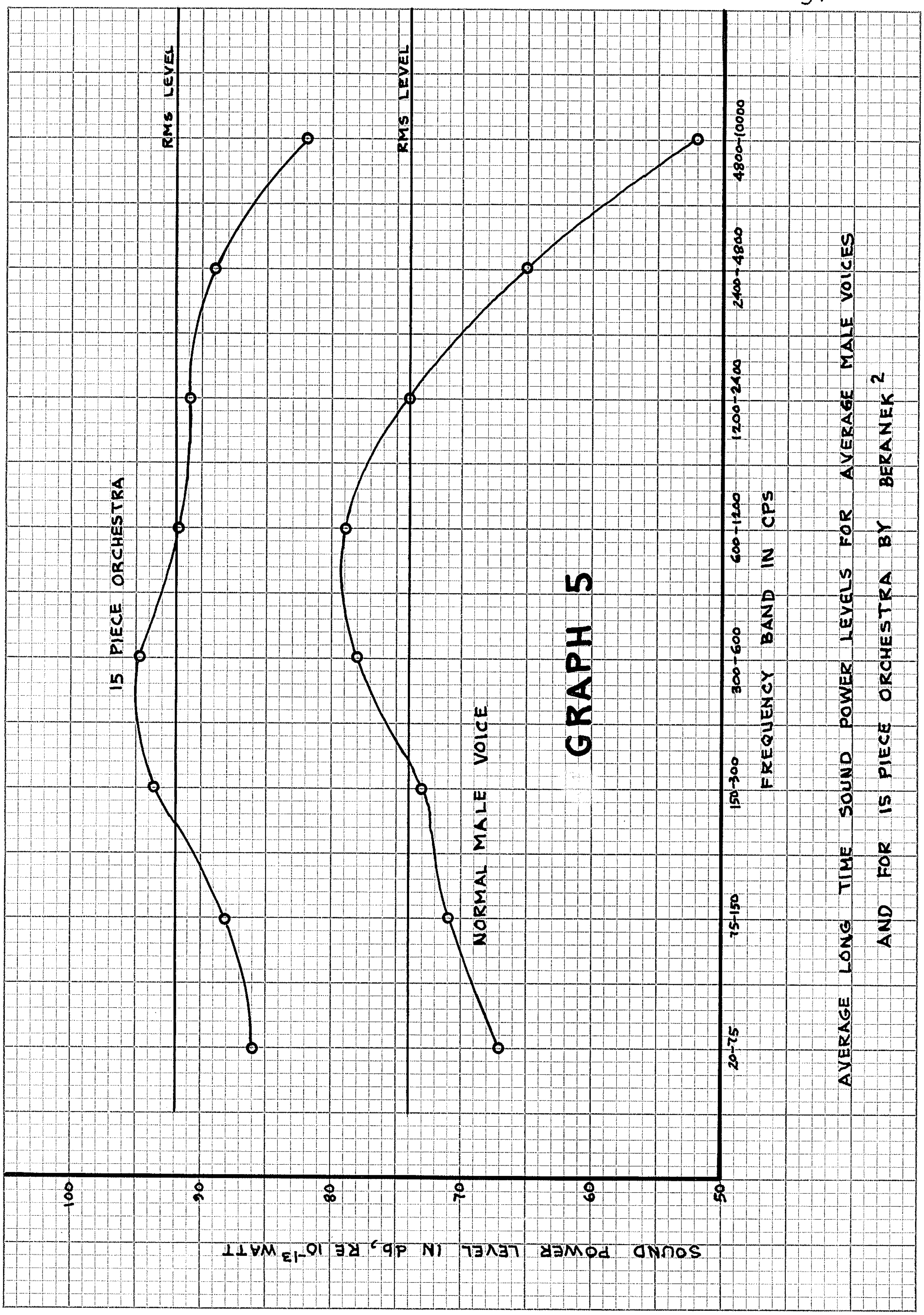
the quality of the musical character as using a lower frequency band.

(3) As given by Fletcher and Munson[3] and shown on GRAPH 2, this band of the noise spectrum had a one to one relationship between loudness and intensity. Thus it closely corresponded to the response of the ear with regard to loudness variations.

(4) A simple, inexpensive noise sensing microphone was available having the required sensitivity and omnidirectional pattern at this band of frequencies.

(5) As shown in GRAPH 5, and as given by Beranek[2], a relatively large amount of noise existed in this band of frequencies and thus only a normal gain amplifier would be required.

(6) As given by Beranek[2] and reproduced in GRAPH 5, this band represented the rms level of both music and male talkers within 1 db. Therefore, this band was not considered overly sensitive to music energy. This was not true for all noise bands. For instance, in the 4,000 to 10,000 cps band, the music level was calculated to be about 10 db below the total rms level. However, the speech level in this band was calculated to be about 22 db below the total rms level. Thus in the case where speech in the form of background noise and background music were present in equal overall effective loudness levels, the



GRAPH 5

AVERAGE LONG TIME SOUND POWER LEVELS FOR AVERAGE MALE VOICES AND FOR 15 PIECE ORCHESTRA BY BERANEK [2] music level would be 22 - 10 = 12 db greater than the speech level in this band. This would have meant that the noise sensing device would have tended to sense the music signal more readily than the speech (noise) signal. Thus, more expensive band rejection and bandpass filters would have been necessary to overcome this 12 db difference.

Accordingly, the slot rejection filter and the bandpass filter shown in FIG. 4 were constructed. These filters were made using available components and were adjusted by trial and error. The frequency response of each filter is shown in GRAPH 6.

After subsequent tests, it was found that this arrangement worked well, being relatively unresponsive to truck noises, and linearily responsive to speech type background noise; e.g., a 10 db increase in noise resulted in a 10 db increase in the gain of the variable gain device.

Tests were conducted to determine the effect of the slot on the quality of the music. The method used was the so called A, B test. This test is considered to be an industry standard[11] when it is desired to determine acoustic differences in similar signals.

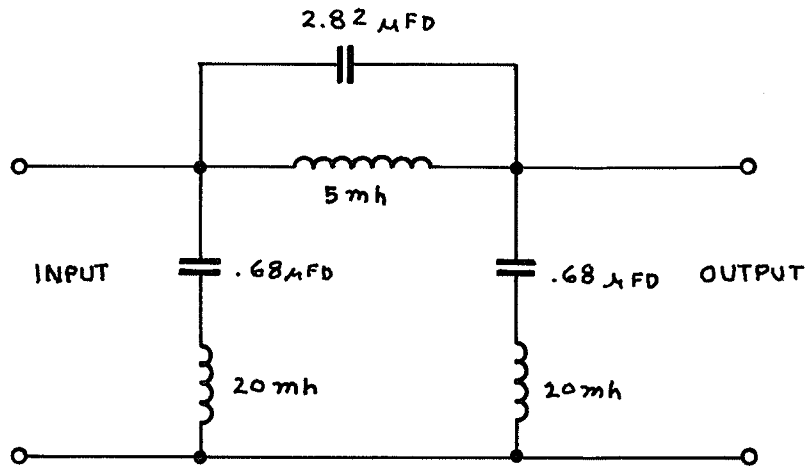


FIG. 4A

MID-FREQUENCY (1150 TO 1650) SLOT

REJECTION FILTER

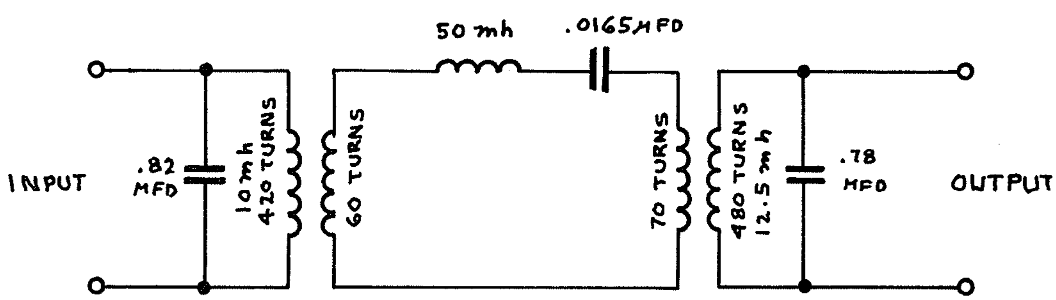
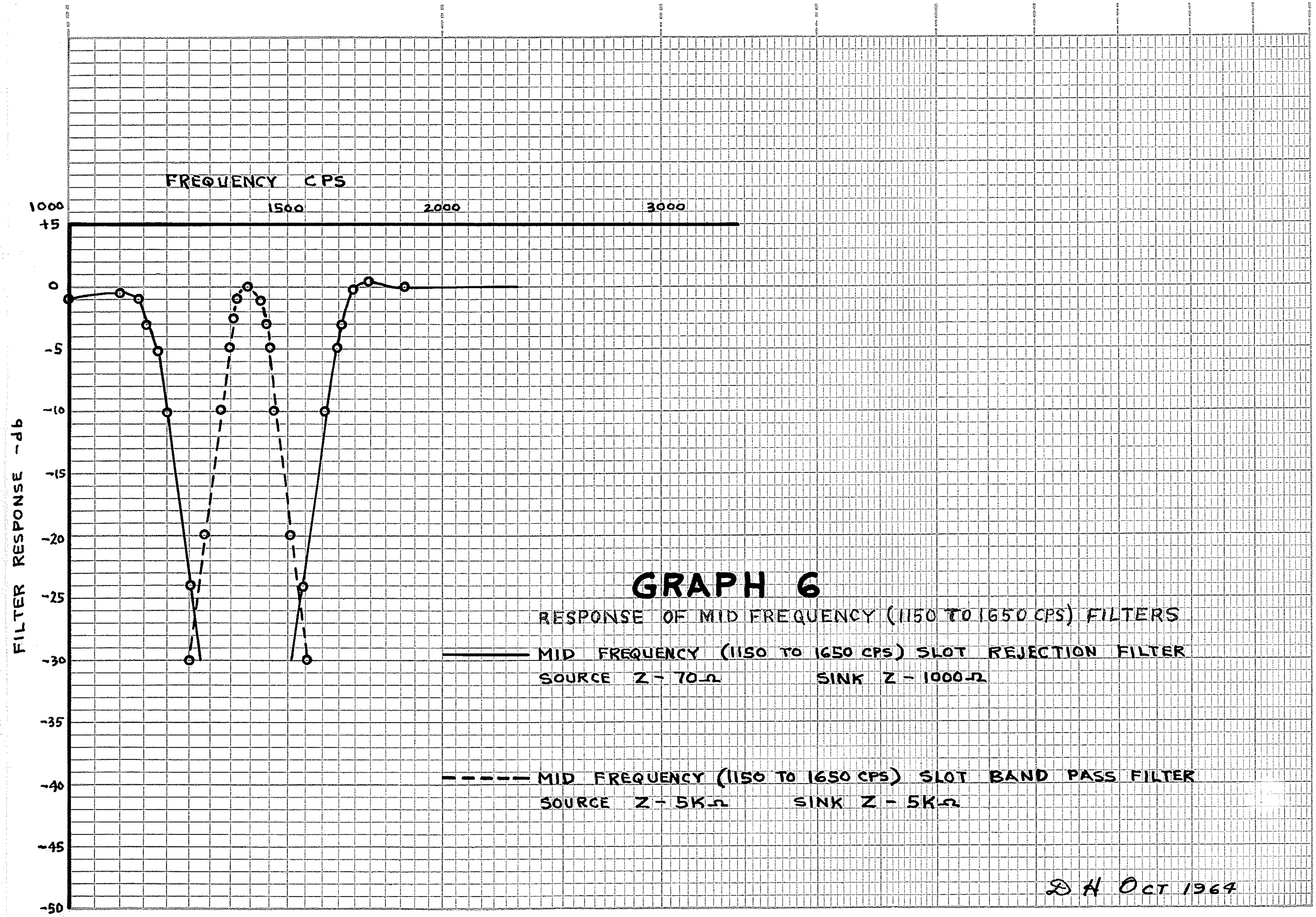


FIG. 4B

MID-FREQUENCY (1150 TO 1650) SLOT

BAND PASS FILTER


GRAPH 6
RESPONSE OF MID FREQUENCY (1150 TO 1650 CPS) FILTERS
MID FREQUENCY (1150 TO 1650 CPS) SLOT REJECTION FILTER
SOURCE Z - 70Ω
SINK Z - 1000Ω
MID FREQUENCY (1150 TO 1650 CPS) SLOT BAND PASS FILTER
SOURCE Z - 5KΩ
SINK Z - 5KΩ
FREQUENCY CPS
1000
1500
2000
3000
FILTER RESPONSE -db
+5
0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
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The test consisted of playing music through a speaker and an amplifier having a quality normal for background music purposes. The music was played first with no slot in the spectrum. Then by means of a switch, the slot was quickly inserted in the music signal. It was arranged so that no level changes occurred. The first switch position was the "A" position, the second the "B" position; thus the reason for calling this method the A, B test. The human ear can readily tell small differences[11] in acoustical qualities providing the time between comparisons is very small.

The same listeners who had given their opinions on previous tests were asked to undertake this A, B test. The tests were conducted in a room considered to be acoustically "live" and to be approximately equivalent acoustically to a typical restaurant or store.

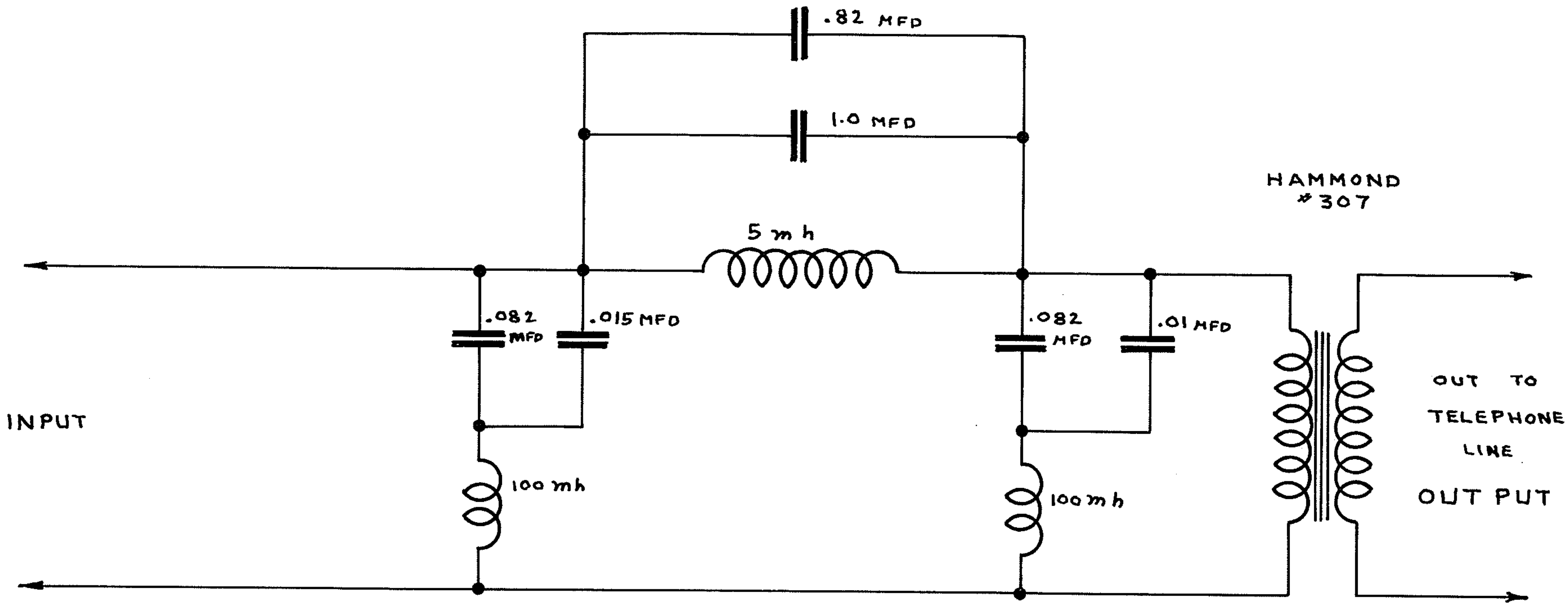
It was the listeners' opinion that the slot caused a serious deterioration in the music signal. In some cases, a whole instrument would seem to disappear. Thus the slot of 1,150 to 1,650 cps was considered to be too large. It was decided to conduct further A, B tests to determine the maximum width of slot that would be acceptable.

For a second choice, it was decided that the slot should be moved up in frequency for two reasons:

(1) so that musical instrument fundamental tones would be less disturbed[8].

(2) so that the frequency of the slot would coincide with the acoustic resonance of a 2 1/4 inch speaker that was to be used as an inexpensive noise sensing microphone. This speaker due to its small size[2], acted as an omnidirectional microphone.

Accordingly, the slot rejection filter shown in FIG. 5, was designed by the "cut and try" method and thence constructed. As shown in GRAPH 8, page 51, the 3 db down points were 1,540 to 1,780 cps; i.e., a 240 cps (1/5 octave) slot. After conducting numerous A, B tests with the same listeners, it was decided that this slot did introduce a change in character of the music, but that it was negligible. The difference between slot and no slot showed up only on certain types of instruments, usually violins, and for most of the musical selections was undetectable. It was thought that any slot, regardless of width, would introduce a degradation in a particular musical selection at a particular instant in the selection. In the case of the slot chosen, the degradation was considered to be barely detectable. When the music was played through the



NOTE:
ALL CONDENSERS SELECTED FOR TUNING TO 1650 CPS.

MID - FREQUENCY
(1540 - 1780 CPS) SLOT
REJECTION FILTER

FIG. 5

slot filter, without reference to the music without the slot filter, listeners could not notice any degradation in the musical character.

It was decided that this slot width and its place in the spectrum was satisfactory. Thence a complete unit was designed and constructed. The unit was designated NOVAC to stand for Noise Operated Volume Automatic Control. The block diagram is shown in FIG. 6, schematic is shown in FIG. 7, and photographs are shown in FIG. 8 and FIG. 9. FIG. 6 and FIG. 7 are located in back cover pocket.

3-4 NOVAC SYSTEM

3-4-1 System Description (Reference FIG. 6)

(1) A music signal, with a slot from 1,540 to 1,760 cps in its spectrum, is introduced at the "Program Input" from either a remote or local program source. Thence, it is amplified in the "Variable Loss Device," and again amplified in the "Amp.," whereupon, it is finally applied to the "Local Sound System."

(2) The "Variable Loss Device" is a device whose loss is controlled by a D.C. signal.

(3) One or more "Noise Sensing Microphones" are located in typically noisy areas and their outputs are connected to the "Noise Amp." via the "Bandpass Filter."

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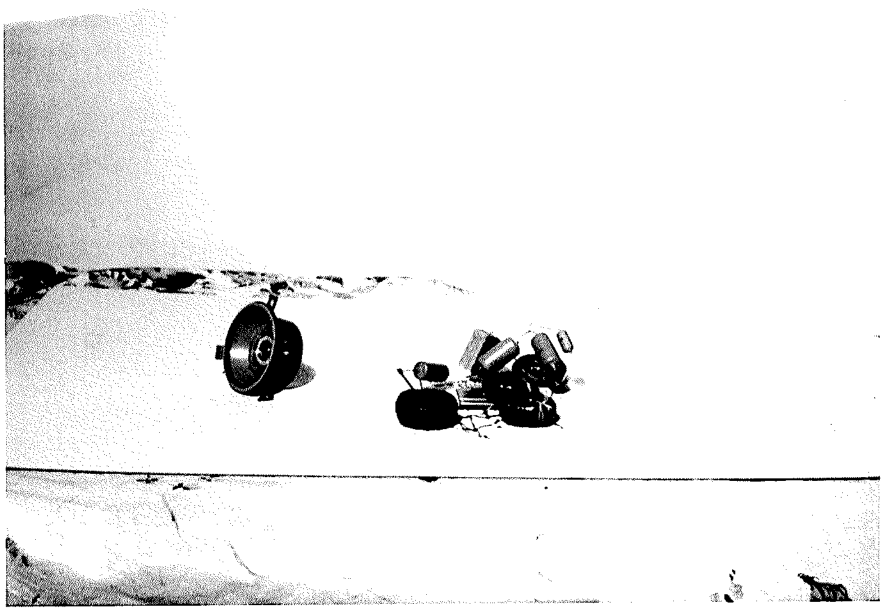
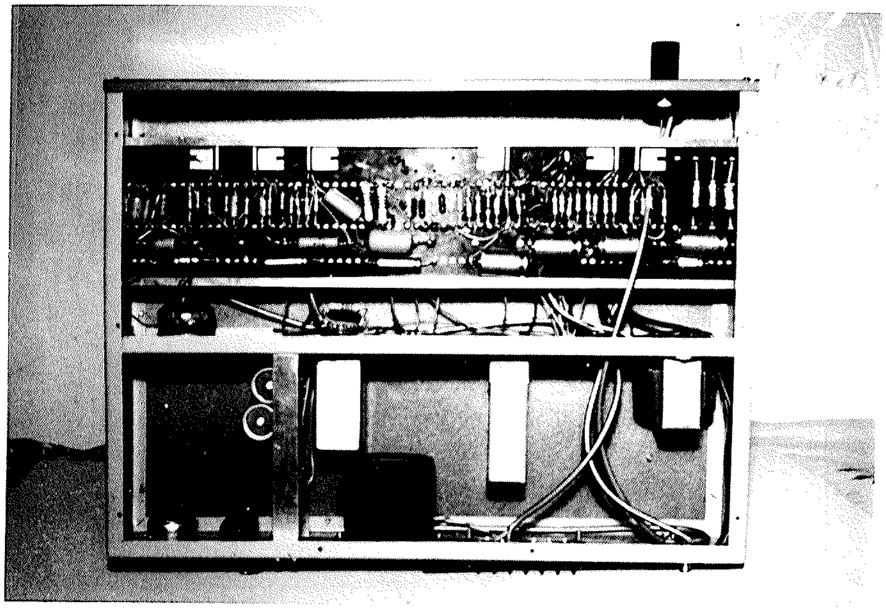
8A Final Slot Rejection Filter (1,540 - 1,780 cps)

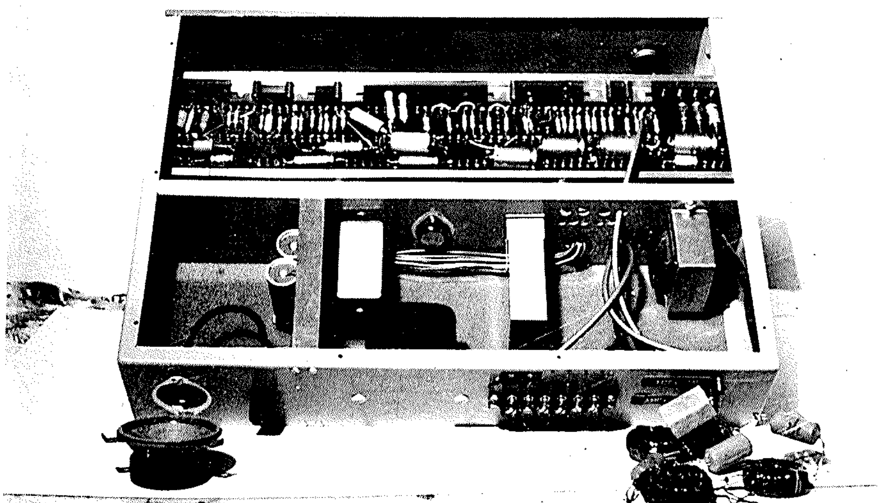


8B Left - Lorenz 2 1/4 inch Speaker Used As Noise Sensing Microphone

Right - Initial Slot Rejection Filter



9A NOVAC Prototype With Top Plate Removed



9B Top - NOVAC Prototype With Top Plate Removed

Left - Lorenz 2 1/4 inch Speaker Used As Noise Sensing Microphone

Right - Initial Slot Bandpass Filter

The "Bandpass Filter" accepts only a narrow band of noise (1,630 to 1,710 cps). This is amplified, rectified, and applied to the "D.C. Amplifier," which controls the "Variable Loss Device." Thus, if the local noise increases, the microphone senses this increased noise and subsequently decreases the loss (i.e., increases gain) through the "Variable Loss Device," thus maintaining a constant signal to noise ratio.

(4) The "Rectifier" is arranged to have a very slow attack time constant (about one minute) but a relatively fast decay time (about 10 seconds).

(5) The "10 db Peak to Rms Noise Clipper" is a device that clips noise peaks greater than 10 db above the average noise.

(6) The long attack time, short decay time, and the "10 db Peak to Rms Noise Clipper" help to prevent sudden bursts of noise from increasing the music signal noticeably via the "Variable Loss Device." Normally, only the average ambient noise produces the D.C. control signal which in turn controls the "Variable Loss Device."

(7) Since the program signal contains negligible energy in the 1,540 to 1,780 cps spectrum, and since this frequency band encompasses that sensed by the "Noise Sensing Microphone," the possibility of feedback from the music signal to the room microphone is greatly reduced.

Thus, in the presence of a normal level of music, the room "Noise Sensing Microphone" senses the background noise. The slot of 1,540 to 1,780 cps is considered to cause a negligible change in the music quality.

(8) Two types of "Local Paging Microphone" circuits are possible.

One type has a band rejection filter in the 1,540 to 1,780 cps bandpass (in the same manner as the program source). This ensures that no adverse positive feedback will occur between the "Noise Sensing Microphone" and the sound system speakers.

The other type has no band rejection filter, but does have an on-off control associated with the "push-to-talk" feature of the "Local Paging Microphone." This control keeps the loss of the "Variable Loss" device fixed during the local message and thus eliminates the possibility of positive feedback between the "Noise Sensing Microphone" and the sound system speakers.

3-4-2 Detailed Circuit Description (Reference FIG. 7)

(1) The "Noise Sensing Microphone," senses room background noise. A bandpass filter, comprising C1, C2, C3, C4, L1, L2, C5, and C6, rejects all noise signals except those that have frequencies in the bandpass from 1,630 to 1,710 cps.

(2) Transistors Q1, Q2, Q3, and Q4 amplify the narrow band of noise in a conventional manner.

(3) Diodes D1 and D2 rectify the signal in a voltage doubler circuit. Resistor R22 applies an initial bias in series with the rectified noise signal voltage. This is required to ensure that D.C. amplifier Q5 is just on the verge of conduction when there is no noise input signal.

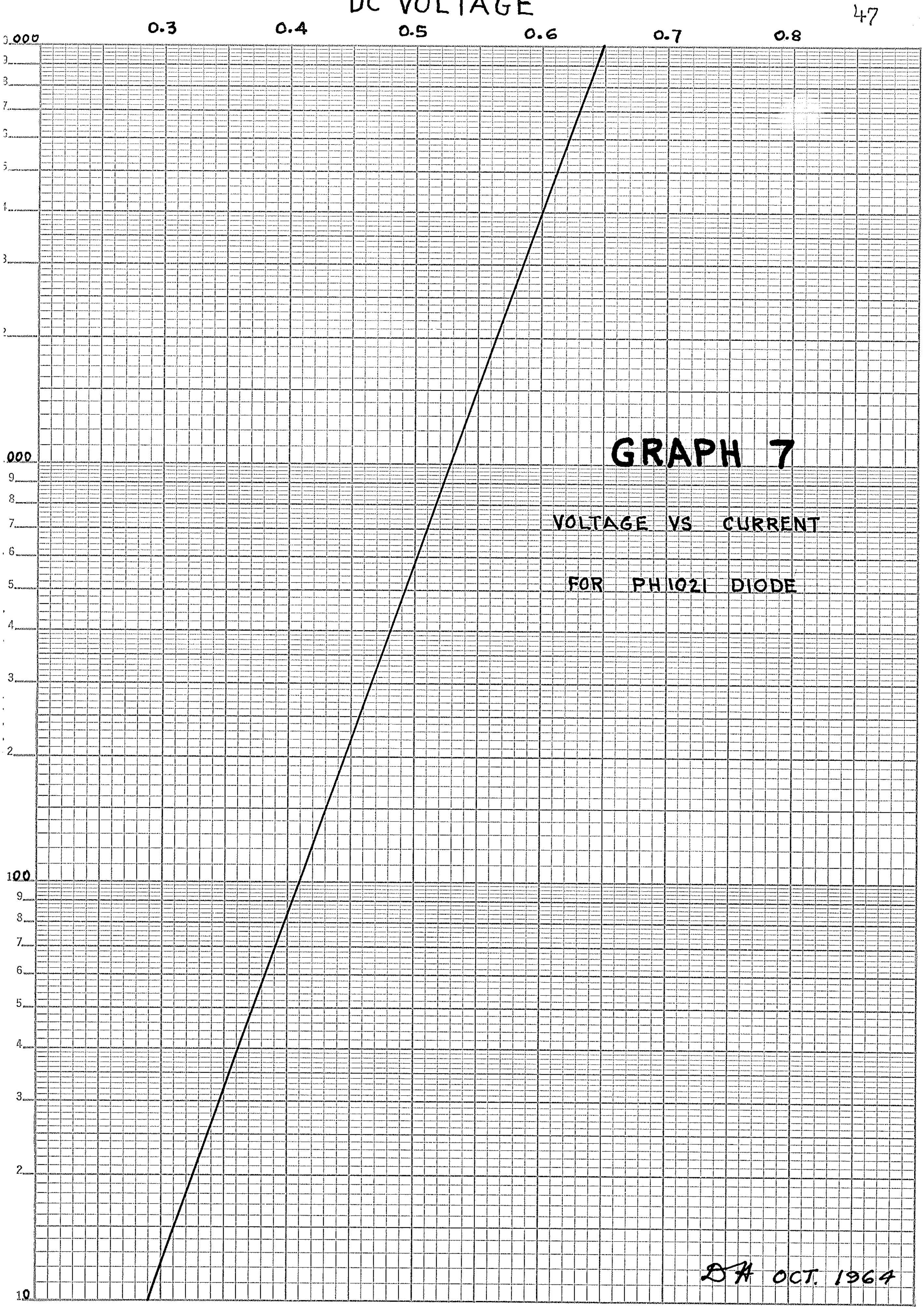
(4) The rectified signal is filtered by condenser C15 and connected via resistors R24 and R25 to the base of D.C. amplifier Q5.

(5) Condenser C16, a large capacity type, causes D.C. variations at the base of Q5 to be of a long time constant (about one minute) when the voltage is rising (increasing noise).

(6) When the noise signal decreases, diode D3, previously not conducting, now conducts and gives a lower time constant (about 10 seconds) for decreasing noise.

(7) The D.C. signal voltage at the emitter of Q5 is applied via resistor R28 to forward conducting diode D7.

(8) Diode D7 is a variable loss device having the characteristic that an <u>increase</u> in the D.C. current through it, will cause a proportionate <u>decrease</u> in its A.C. resistance. The voltage, current relationship is shown in GRAPH 7.


DC VOLTAGE
0.3
0.4
0.5
0.6
0.7
0.8
GRAPH 7
VOLTAGE VS CURRENT
FOR PH1021 DIODE
DH OCT. 1964

(9) The diode D7 is in the emitter circuit of Q7. The program signal enters via "Program Input" socket, to the volume control P1, through the coupling capacitor C19 and thence to the base of Q7.

(10) If the D.C. current through D7 increases, its A.C. impedance <u>decreases</u> and the gain of transistor Q7 increases, due to the reduction in emitter negative feedback.

(11) Thus an increase of the noise signal will cause an <u>increase</u> in the gain of Q7. This is relatively linear over a 30 db range.

(12) The program signal is finally amplified by transistor Q8, applied to the "Program Output" socket, and thence to the local sound system.

(13) A D.C. signal is also produced at the collector of Q5 and is thence applied to the base of npn transistor Q6.

(14) Diodes D5 and D6 are in the circuit for temperature compensation of the D.C. amplifier consisting of Q5 and Q6.

(15) The amplified D.C. signal appears on the collector of Q6. Since the D.C. signal has undergone two inversions from the base (also emitter) of Q5, the voltage at the collector of Q6 is in phase. That is, an

increase in the D.C. voltage at the base of Q5 will cause an amplified increase in the D.C. voltage at the collector of Q6.

(16) This amplified D.C. voltage is applied to the junction of R24 and R25 via diode D4.

(17) If, due to an abnormally loud noise, a sudden peak of noise voltage appears at the output of the rectifier, this voltage will not unduly charge up condenser C16, as it will be clipped beyond a certain level established by the D.C. voltage on the collector of Q6. Resistors R24, R25, R26, R30, R29 and R32 determine this level. The circuitry is designed so that only noise signals greater than 10 db in level above the average are clipped.

(18) Relay K is under the external control of a local paging microphone "push-to-talk" switch. When the microphone switch is pushed, relay K operates and performs two functions. Firstly, it shorts, via contact K2, the "Program Output" and thus mutes the music signal. This prevents the music from interfering with the spoken announcement. Secondly, it opens the ground connection to capacitor C16 and thus keeps it from discharging. This effectively holds constant the D.C. voltage on C16. Thus, when the microphone switch is released, the D.C. signal voltage will be the same as before and no noticeable change in program level will result.

3-5 SPECIAL DESIGN FEATURES

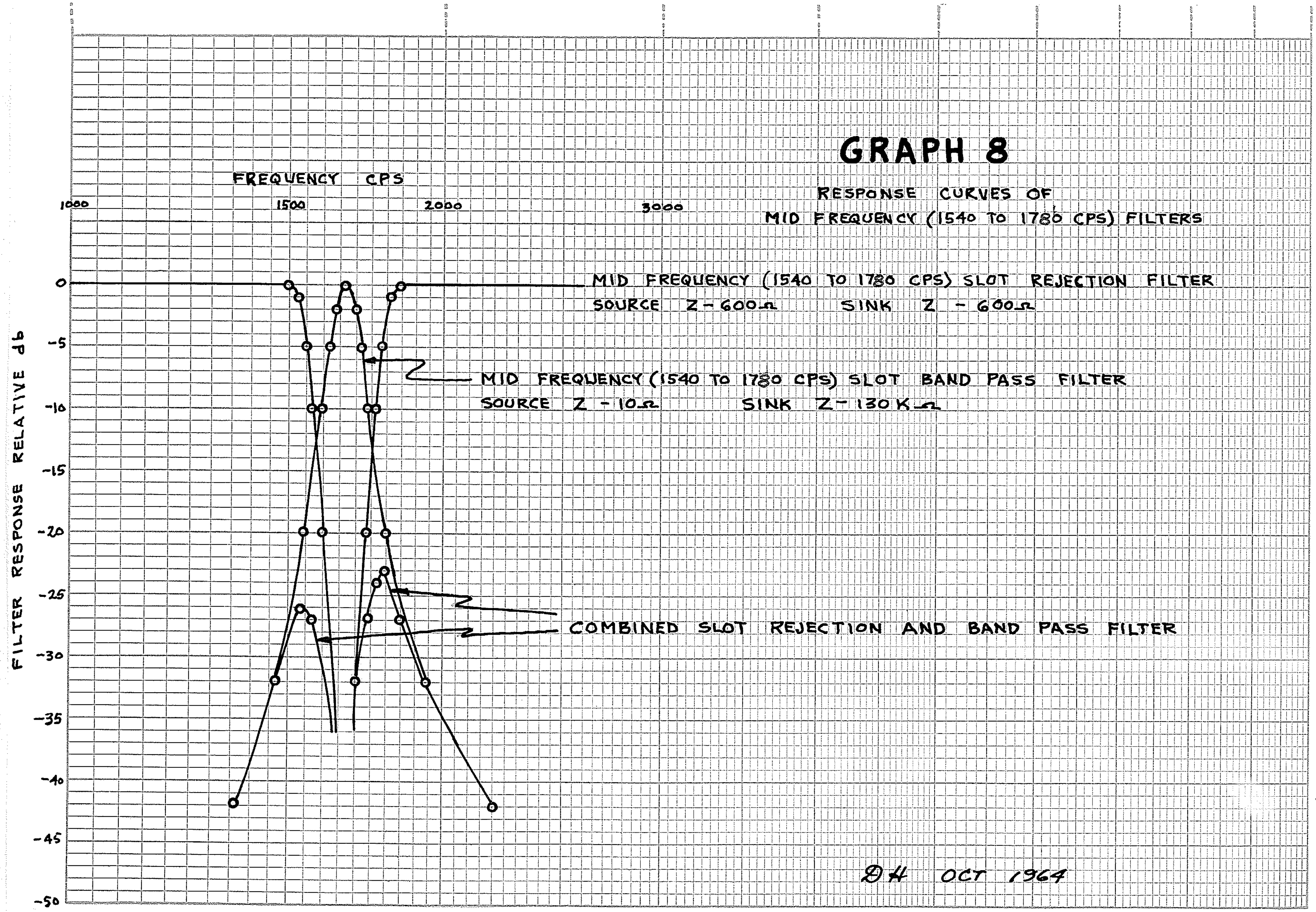
(1) The noise bandpass filter consists of a single critically coupled pair of tuned circuits having characteristics similar to those found in the I. F. amplifier of an ordinary radio set. After testing this filter in conjunction with the rejection filter, it was considered that this circuit was satisfactory in its response characteristic and was reasonable in cost. Ferrite pot cores were used for the inductors[13]. These cores were found ideal for filter applications because:

(i) their inductance could be continuously adjusted over a $\pm 7\%$ range.

(ii) the bobbins used with the cores could easily be wound for almost any value of inductance.

(iii) they were low in cost.

Polystyrene condensers were chosen for use in the filter because of their capacitance stability with time and also because their temperature coefficient was found to be approximately the inverse of the ferrite pot cores. This tended to make the filter characteristic independent of temperature. The response is shown in GRAPH 8 along with the combined response of the slot bandpass and the rejection filter. As can be seen, the combined response has two minor lobes on either side of the bandpass filter. This is the result of the imperfections of the rejection


GRAPH 8
RESPONSE CURVES OF
MID FREQUENCY (1540 TO 1780 CPS) FILTERS
FREQUENCY CPS
1000
1500
2000
3000
FILTER RESPONSE RELATIVE db
0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
MID FREQUENCY (1540 TO 1780 CPS) SLOT REJECTION FILTER
SOURCE Z - 600Ω SINK Z - 600Ω
MID FREQUENCY (1540 TO 1780 CPS) SLOT BAND PASS FILTER
SOURCE Z - 10Ω SINK Z - 130KΩ
COMBINED SLOT REJECTION AND BAND PASS FILTER
DH OCT 1964

and bandpass filters. These lobes represent the response of the noise sensing channel to the music signal. The lobe on the left side is 26 db down and has a bandwidth of 90 cps. The lobe on the right side is 23 db down and has a bandwidth of 80 cps. The slot bandpass filter has a bandwidth of 80 cps. By combining the bandwidths and levels of the two side lobes on an rms basis and comparing this combined level and bandwidth with the slot bandpass filter level and bandwidth, it was calculated that the music signal was rejected by 21 db in comparison to the background noise signal. This meant that the noise sensing channel would discriminate between the music signal and background noise signal in that the noise signal would be favoured by 21 db over the music signal.

(2) The noise clipper consisting of Q5, Q6, diode D4, and associated components, was found necessary to prevent single, widely spaced sudden noises, from unduly causing an increase in gain in the NOVAC. The clipper functions as follows:

Reference FIG. 7

(1) Assume background noise is sufficient to produce a 6 volt D.C. signal at A and thus B.

(2) The Q5, Q6, amplifier combination has a gain of 1.5 and thus at C the D.C. voltage will be:

$$1.5 \times 6 = 9 \text{ volts.}$$

(3) Now assume noise increases momentarily to produce a 16 volt signal at A.

(4) The voltage at D will be

$$\left(\frac{39K}{39K + 75K} \text{ X (voltage at A - voltage at B)} \right)$$

+ Voltage at B

$$= \left(\frac{39}{39 + 75} \text{ X } (16 - 6) \right) + 6$$

$$= \underline{\underline{9.5}} \text{ volts.}$$

(5) This will just cause diode D4 to conduct.

(6) If the voltage at A were to increase to greater than 16 volts, the voltage at D would be prevented from following this increase since the diode D4, in conducting, would hold the voltage at D at 9.5 volts. This would prevent the voltage at B from rising. Since the voltage at B controls the NOVAC gain, then the noise voltage at A that has risen above 16 volts has had no further effect on the NOVAC gain than a noise voltage of 16 volts, and thus the effect of this increased voltage has been "clipped." The voltage ratio of $\frac{16}{6}$ is approximately 10 db. That is, clipping action takes place when the noise peaks are greater than 10 db above the average ambient noise. If the voltage at A

were to remain below 16 volts, no clipping would occur. The clipping action depends on the voltage at C which is controlled by the voltage at A. Thus, regardless of the voltage at B, a 10 db clipping level above average noise will be maintained. The action of the clipper was examined on an oscilloscope for approximately ten minutes during which time restaurant noise was played back from a tape recorder. It was observed that the normal average ambient noise was seldom clipped but that large sudden noises were. If the clipping point was made less than 10 db above the average, excessive clipping of the average ambient noise was observed. If the clipping point was made greater than 10 db above the average then the sudden noises were not clipped as much as they could be.

CHAPTER FOUR

FINAL TEST RESULTS OF COMPLETED NOVAC DEVICE

In order to determine the limitations and the performance of the NOVAC prototype device, a number of tests were performed. Refer to FIG. 7, located in the back cover pocket, for reference to the points at which measurements were taken.

4-1 NOVAC GAIN VS BACKGROUND NOISE LEVEL

The purpose of these tests was to determine the relationship between the background noise level and the gain of the NOVAC gain which in turn controlled the music level.

(1) Taped background noise from the Embassy Grill restaurant was played through an amplifier and speaker at different noise levels. For each measurement, the same part of the tape was used. This meant that the only difference in the noise signal for each measurement was the level of the noise and not its character. The NOVAC gain was measured at each noise level by applying a constant 1,000 cps tone to the music input and measuring the tone output level.

TABLE 4

NOVAC GAIN VS BACKGROUND NOISE LEVEL - NOISE FOR EACH READING ALWAYS THE SAME PART ON TAPE OF RECORDED NOISE

Background Noise Level (db above ref.)	NOVAC Gain (relative db)
79	0
74	-5.0
69	-9.8
64	-14.7
59	-19.6
55	-24.3
54	-29.0

These results were plotted on GRAPH 9.

(2) Test (1) was repeated except the same part of the taped noise was not repeated for each measurement. Instead, for each reading, the background noise was different in character from the preceding reading; i.e., a different part of the recorded tape was used.

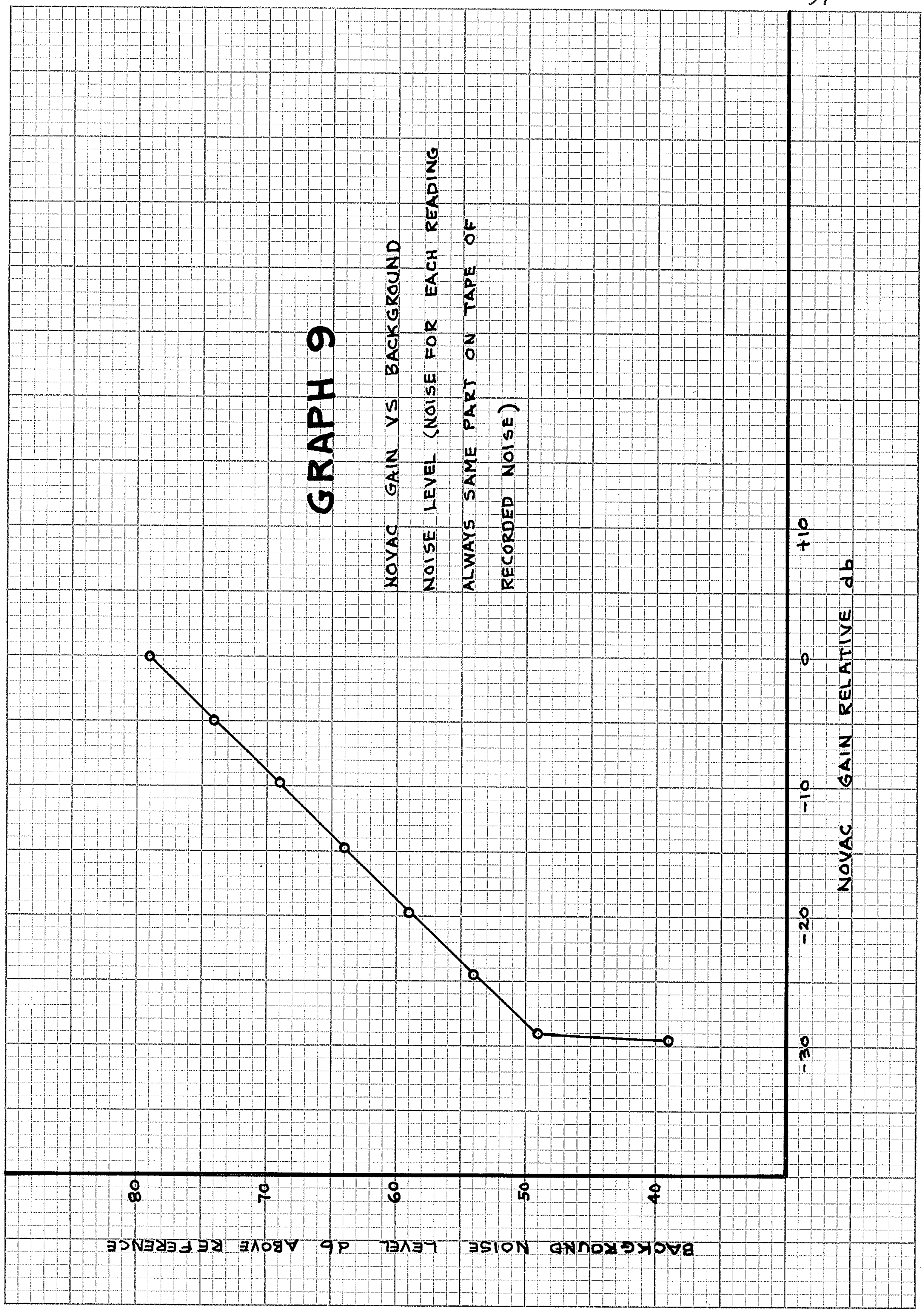

GRAPH 9
NOVAC GAIN VS BACKGROUND NOISE LEVEL (NOISE FOR EACH READING ALWAYS SAME PART ON TAPE OF RECORDED NOISE)
NOVAC GAIN RELATIVE db
-30
-20
-10
0
+10
BACKGROUND NOISE LEVEL db ABOVE REFERENCE
80
70
60
50
40

TABLE 5

NOVAC GAIN VS BACKGROUND NOISE LEVEL - NOISE FOR EACH READING ALWAYS DIFFERENT PART ON TAPE OF RECORDED NOISE

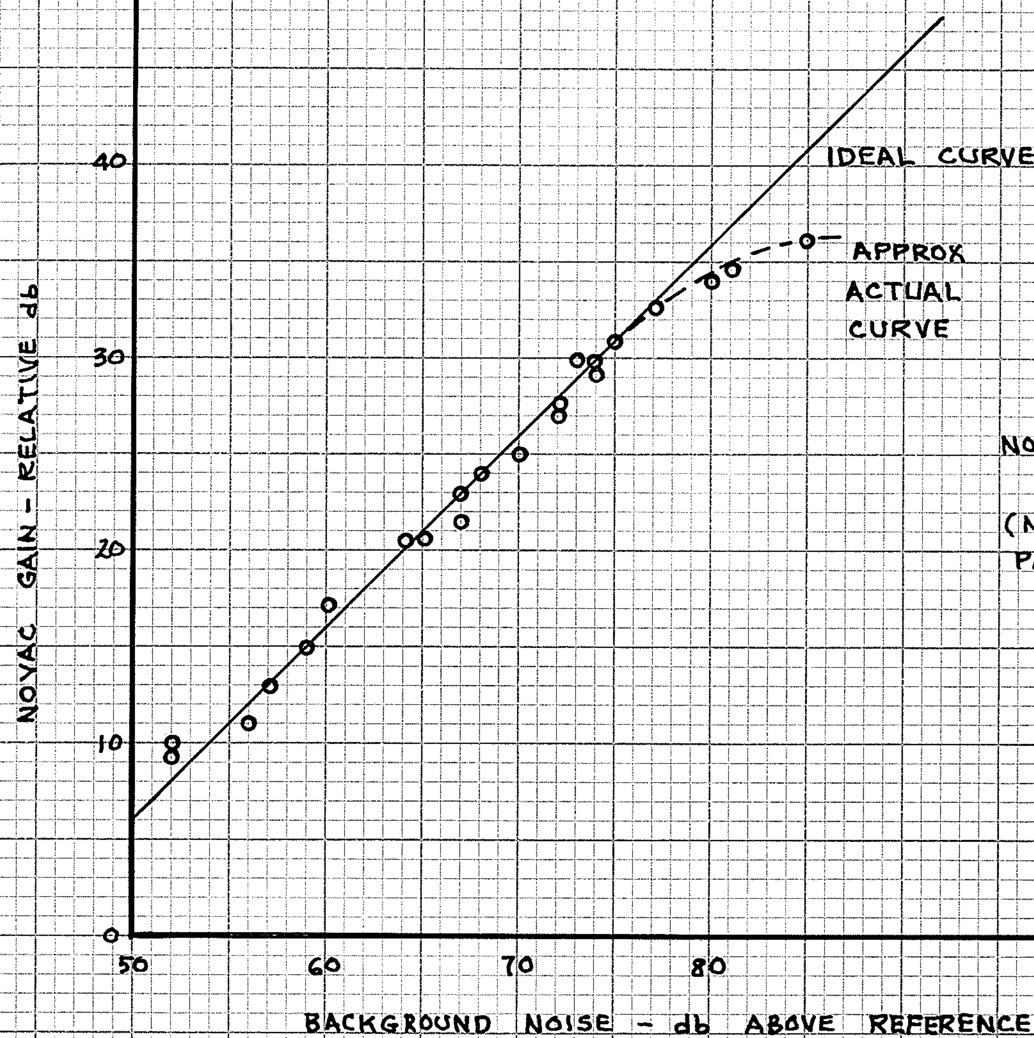
Background Noise Level (db above ref.)	NOVAC Gain (relative db)
85	36.0
81	34.7
80	34.0
77	32.7
75	31.0
74	29.2
74	29.7
73	30.0
72	27.5
72	27.0
70	25.0
68	24.0
67	23.0
67	21.5
65	20.5
64	20.5
60	17.0
57	13.0
56	11.0
52	10.0
52	9.3

These results were plotted on GRAPH 10.

GRAPH 10

NOVAC GAIN VS BACKGROUND NOISE LEVEL

(NOISE FOR EACH READING ALWAYS DIFFERENT PART ON TAPE OF RECORDED NOISE)



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(3) Test (1) was repeated except a steady 1,680 cps tone was used for the noise input instead of taped background restaurant noise.

TABLE 6

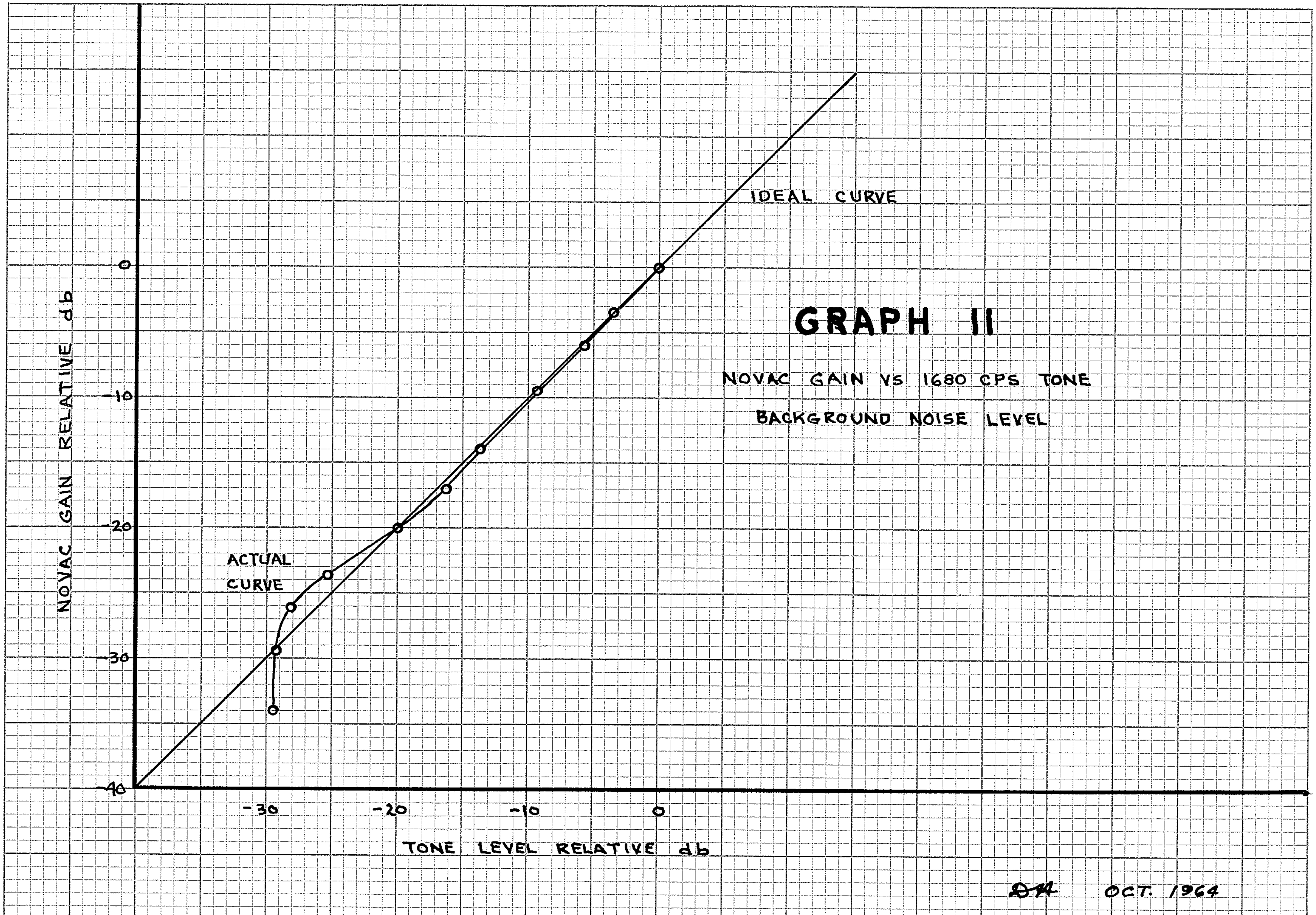
NOVAC GAIN VS 1,680 CPS TONE BACKGROUND NOISE LEVEL

Tone Level (relative db)	NOVAC Gain (relative db)
0	0
-3.5	-3.2
-6.0	-5.6
-9.5	-9.3
-14.0	-13.7
-17.0	-16.6
-20.0	-20.0
-23.5	-25.2
-26.0	-28.2
-29.5	-29.3
-34.0	-29.5

These results were plotted on GRAPH 11.

4-2 NOVAC GAIN VS D.C. CONTROL VOLTAGES

The purpose of this test was to correlate the gain of NOVAC with the D.C. control voltages in the music variable gain amplifier. This information was


GRAPH II
NOVAC GAIN VS 1680 CPS TONE
BACKGROUND NOISE LEVEL
IDEAL CURVE
ACTUAL CURVE
NOVAC GAIN RELATIVE db
TONE LEVEL RELATIVE db
0
-10
-20
-30
-40
OCT. 1964

then used in subsequent tests.

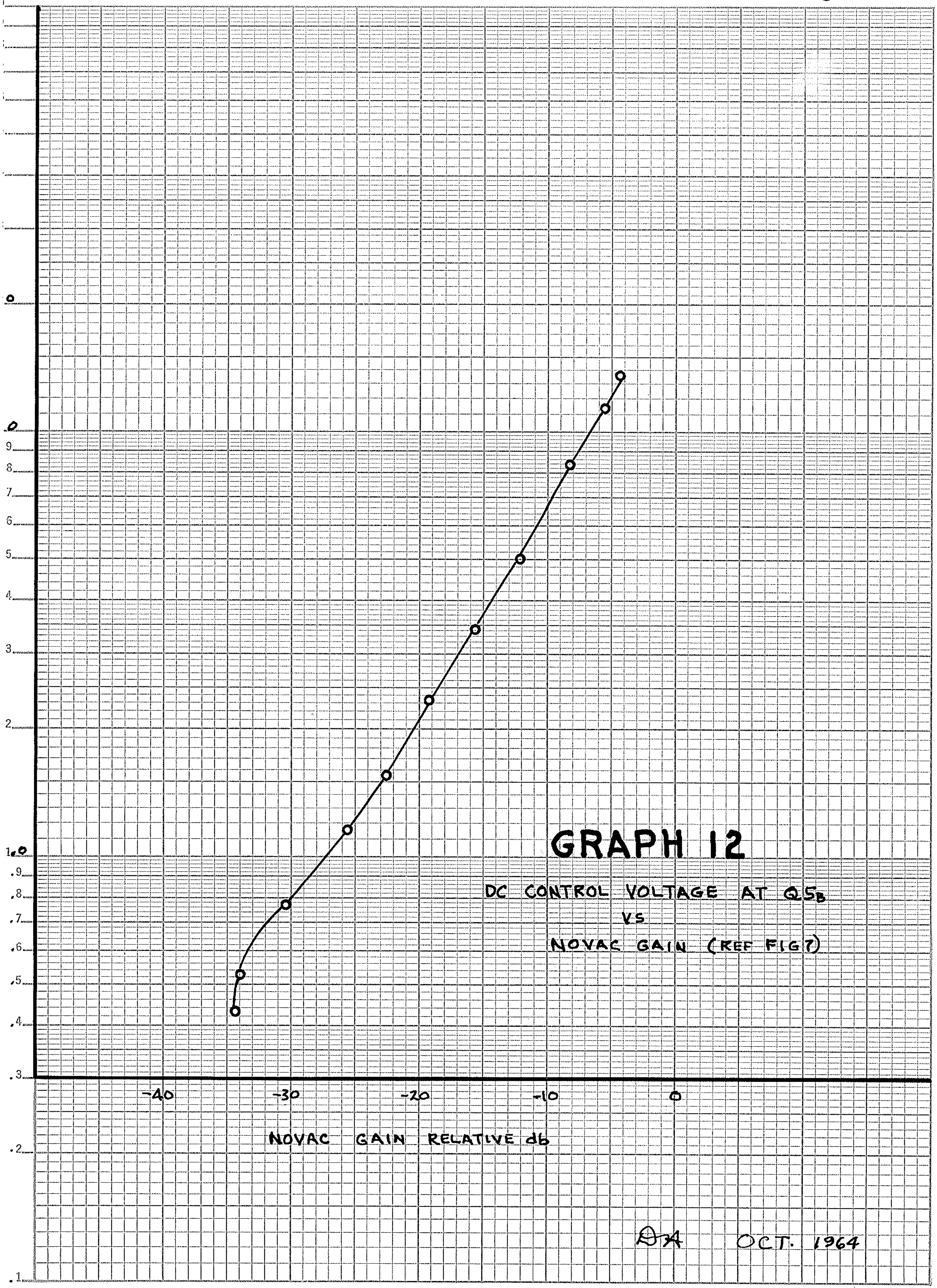
The NOVAC gain was controlled by means of a steady 1,680 cps tone. The D.C. control voltages were recorded for various NOVAC gains.

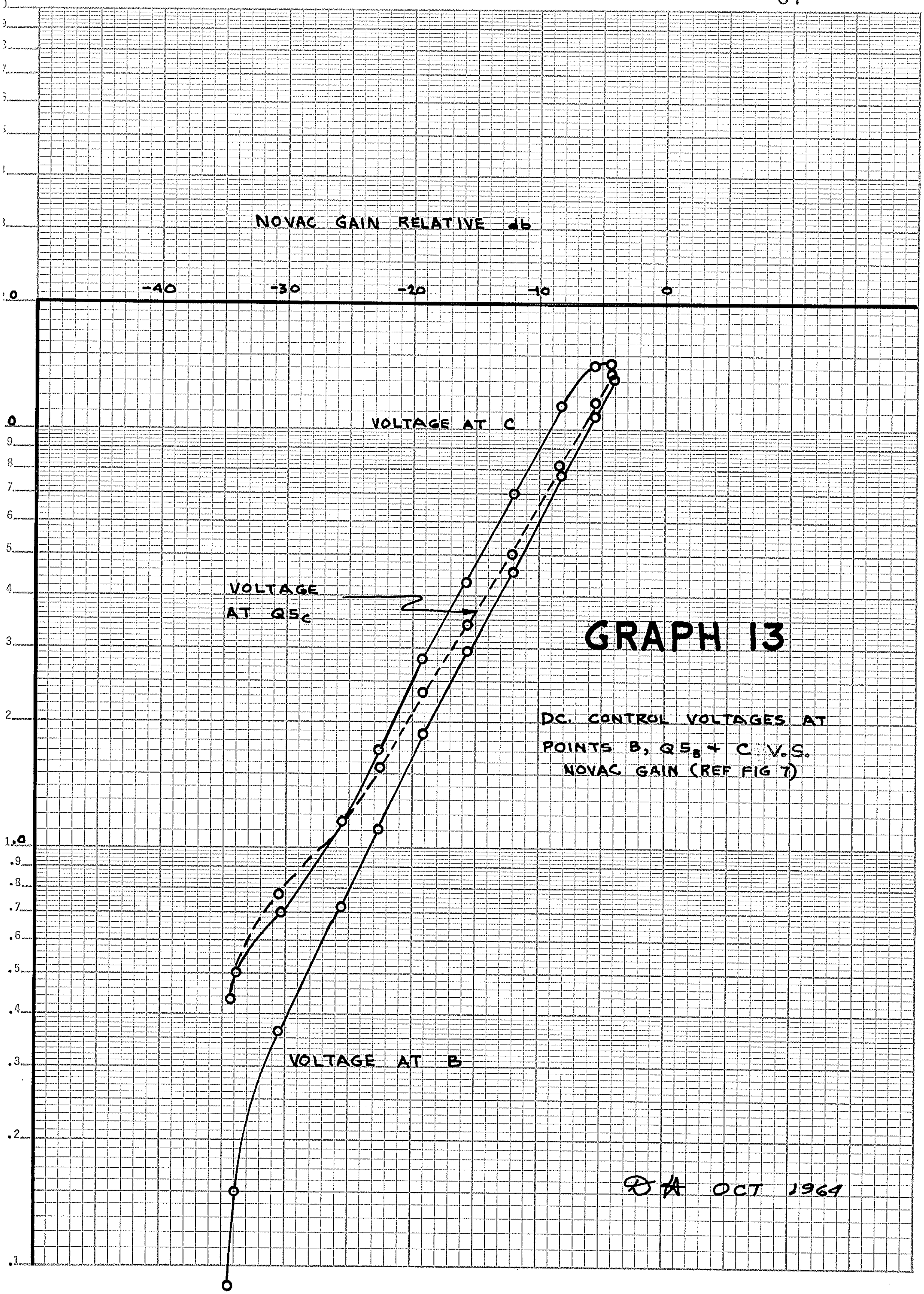
TABLE 7

NOVAC GAIN VS D.C. CONTROL VOLTAGES

NOVAC Gain (relative db)	Control Voltage (D.C.) $Q5_B$	$Q5_E$	$Q6_C$
-4.3	13.6	13.1	14.2
-5.6	11.4	10.9	14.2
-8.4	8.5	7.8	11.4
-12.1	5.0	4.56	7.00
-15.8	3.4	2.95	4.30
-19.3	2.32	1.87	2.80
-22.7	1.53	1.10	1.70
-25.6	1.15	.72	1.16
-30.6	.77	.36	.70
-34.0	.52	.15	.50
-34.3	.43	.08	.45

These results are plotted on GRAPHS 12, 13.


GRAPH 12
DC CONTROL VOLTAGE AT Q5B
VS
NOVAC GAIN (REF FIG 7)
NOVAC GAIN RELATIVE db
-40
-30
-20
-10
0
OCT. 1964


NOVAC GAIN RELATIVE db
-40
-30
-20
-10
0
VOLTAGE AT C
VOLTAGE AT Q5c
VOLTAGE AT B
1.0
GRAPH 13
DC. CONTROL VOLTAGES AT POINTS B, Q5B + C V.S. NOVAC GAIN (REF FIG 7)
OCT 1964

4-3 EFFECT OF MUSIC LEVEL ON NOVAC GAIN

If the slot rejection and bandpass filters were ideal, the music signal would be completely blocked from entering the noise sensing channel. This was because the two filters were designed to complement each other. With this arrangement, the NOVAC device would control the music level regardless of how large the music signal to noise ratio was. However, the filters that were used were constructed as economically as possible and thus were relatively imperfect. Thus the music signal did, to a certain extent, enter the noise sensing channel via the bandpass filter. In doing so, it introduced a signal which the noise amplifier considered as noise. This signal was amplified, and as a result caused the music signal to increase in the same manner as an increasing noise signal would normally do. The increased music signal in turn produced more signal in the noise sensing band which caused a further gain increase. This was effectively a positive feedback signal from the music channel to the noise sensing channel, and then back to the music channel.

In order to determine the effect of music level on the NOVAC gain, the following test was performed.

(1) A 1,680 cps tone to simulate a steady background noise was introduced into the NOVAC noise sensing microphone with sufficient level to produce 2.1 volts D.C. at $Q5_B$.

(2) Music was played through the NOVAC over a range of input levels.

(3) For each input level, the maximum D.C. voltage that appeared during a musical selection at $Q5_B$ was recorded for a number of musical selections.

TABLE 8

MUSIC LEVEL VS CHANGE IN D.C. CONTROL VOLTAGE

Input Music Signal Level of NOVAC (relative db) Note 1	Maximum D.C. Level at $Q5_B$ During each Musical Selection (Note 2)							
	Musical Selections							Avg.
	#1	#2	#3	#4	#5	#6	#7	
+5	2.3	2.3	2.25					2.28
+10	2.4	2.35	2.4	2.3				2.38
+15	2.8	2.6	2.6	2.9	3.0	2.6	3.5	2.86
+20	4.0	4.0	10.0*	4.4	3.5			3.9
+25	11.0	13.5	12.0	13.0				12.4

*Accordian Selection - This was neglected in averaging. This unusually large result occurred only at the fairly high input music level of +20. Also, this type of selection was found infrequently in the normal background music repertoire. It was thought that under normal music levels, this large increase in level would not normally be expected.

Note (1) By measurement, an input level of +5 dbm of music corresponded to an acoustical music level of 65 db above reference.

(2) By measurement, 2.1 D.C. volts at $Q5_B$ corresponded to an acoustical noise level of 63 db above reference.

If there had been no effect of the music signal on the noise sensing channel, then the D.C. control voltage at $Q5_B$ would not have changed, as this voltage represented the NOVAC gain as determined by the background noise level. Any change must have been caused by the music signal entering the noise channel due to the fact that the filters used were not ideal.

This meant that the music signal tended to increase its own level via the noise sensing microphone and variable gain amplifier. Table 9 was calculated from the results of Table 8 as follows:

Referring to Table 8, consider the first result; i.e., Music Signal of +5. As per Note (1), this corresponded to 65 db of noise above reference. As per Note (2) 2.1 volts D.C. at $Q5_B$ corresponded to 63 db of noise above reference. This gave an initial 65 - 63 = +2 db music signal to noise ratio <u>before</u> the music tended to change its own level.

Referring to GRAPH 12, the difference of 2.1 volts and 2.28 volts (these readings obtained from Table 8) was 0.6 db difference in NOVAC gain.

Thus Table 9 shows that for a +2 db music signal to noise ratio, the music increased its own level by 0.6 db. This calculation was repeated for each of the remaining measurements.

TABLE 9

MUSIC LEVEL VS CHANGE IN NOVAC GAIN

Music Signal to Noise Ratio (db) (Measured Before Change in Gain of NOVAC Due to Music Level)	Avg. Increase in Music Level (db) Due to Music Level
+2	.6
+7	1.0
+12	2.4
+17	5.4
+22	16. (runaway)

4-4 REJECTION OF SLOT, BANDPASS AND REJECTION FILTERS TO MUSIC

The purpose of this test was to note the efficiency with which the slot rejection and bandpass filters tended to keep the music signal out of the noise sensing channel. To determine this, the D.C. level at $Q5_B$ was noted for various musical selections first with the slot filter and then without the slot filter. In this test, the music signal was not sent through the NOVAC device. Instead the music signal was produced by means of a normal amplifier and the music level was adjusted by means of its gain control. Thus, the gain of the NOVAC had no effect on the music signal level.

TABLE 10

EFFECT OF MUSIC ON D.C. CONTROL VOLTAGE WITH SLOT FILTER
(MUSIC LEVEL: 85 DB ABOVE REF.)

Musical Selection	Largest D.C. Voltage at $Q5_B$ Obtained During Musical Selection
#1	5.6
#2	6.0
#3	4.0
#5	4.5
#6	4.5
Avg.	5.0

TABLE 11

EFFECT OF MUSIC ON D.C. CONTROL VOLTAGE WITHOUT SLOT FILTER (MUSICAL LEVEL: 70 DB ABOVE REF.)

Musical Selection	Largest D.C. Voltage at $Q5_B$ Obtained During Musical Selection
#1	3.0
#2	6.2
#3	4.5
#4	3.8
#5	4.5
Avg.	4.4 Volts D.C.

From Tables 10 and 11, the degree by which the slot rejection and bandpass filters kept the music signal from entering the noise sensing channel was calculated as follows:

$$a = (b - c) + (d - e)$$

where a is the average rejection of music signal due to the slot rejection and bandpass filters.

b is the music level used to derive Table 10.

c is the music level used to derive Table 11.

d is the relative gain of NOVAC associated with average D.C. control voltage of Table 10 (this determined from GRAPH 12, page 63).

e is the relative gain of NOVAC associated with average D.C. control voltage of Table 11 (this determined from GRAPH 12, page 63).

In this case, a, is:

$$= (85 - 70) + (-12.3 - (-13.3))$$
$$= 15 + 1$$
$$= 16 \text{ db average}$$

4-5 CORRELATION BETWEEN ACOUSTIC BACKGROUND NOISE AND D.C. VOLTAGE AT $Q5_B$

In order to have a reference between the background noise and the D.C. control voltage at $Q5_B$, the voltage at $Q5_B$ was measured for a number of noise levels.

TABLE 12

BACKGROUND NOISE VS D.C. CONTROL VOLTAGE AT $Q5_B$

Background Noise Level (db above ref.)	D.C. Control Voltage at $Q5_B$
60	1.4
60	1.6
61	1.5
63	2.0
60	1.5
59	1.4
64	1.9
67	2.7

When the D.C. voltages were corrected to 60 db noise above reference, (referring to GRAPH 12 and using as a first approximation that 60 db above reference was equal to 1.4 volts) the following was obtained:

TABLE 13

CORRECTED BACKGROUND NOISE VS D.C. CONTROL VOLTAGE AT $Q5_B$

Background Noise Level (db above ref.)	Corrected D.C. Control Voltage at $Q5_B$
60	1.4
60	1.6
60	1.25
60	1.4
60	1.5
60	1.55
60	1.20
60	1.25
Avg.	1.4 $\pm$.25 volts

Therefore, from this limited data, 60 db noise above reference produced 1.4 $\pm$.25 D.C. control volts at $Q5_B$.

4-6 DISTORTION OF NOVAC TO MUSIC SIGNAL

In order to determine to what degree the variable gain amplifier distorted the music signal, various tone frequencies were passed through the NOVAC device. The input level of these tones was varied and also the D.C. control voltage was varied in order to determine their effect on the distortion of the tones. The results were as follows:

TABLE 14

DISTORTION OF NOVAC AT VARIOUS INPUT LEVELS AND CONTROL VOLTAGES

Input Frequency (cps)	Input Level (mv)	Control Voltage at $Q5_B$	Output Level (volts)	Harmonic Distortion %
600	2.0	-4.4	.4	1.5
600	6.0	-4.4	1.2	2.7
1,000	8.0	-1.9	.67	2.9
10,000	8.0	-1.9	.53	2.9
100	8.0	-1.9	.64	2.8
1,000	8.0	-4.0	1.35	3.5
1,000	4.0	-4.0	.67	1.7
1,000	1.3	-12.0	.67	1.25
1,000	2.6	-12.0	1.35	1.15
1,000	3.9	-12.0	2.0	1.5
10,000	3.9	-12.0	1.7	2.1
5,000	3.9	-12.0	1.85	2.4
5,000	2.0	-12.0	.95	1.35

Note: Residual distortion of oscillator was .3%.

4-7 CHARGE AND DISCHARGE TIMES OF CONTROL CIRCUIT OF NOVAC

If the background noise were to increase or decrease rapidly and stay at the new noise level, then the NOVAC gain would change in order to keep the same music to noise ratio. However, the change would not be instantaneous due to the charge and discharge time constants of the circuits used to control the gain of the NOVAC. Tables 15, 16 and 17 show how the gain changed with time when the noise level was varied.

4-7-1 Time vs Gain for an Increasing Noise Signal

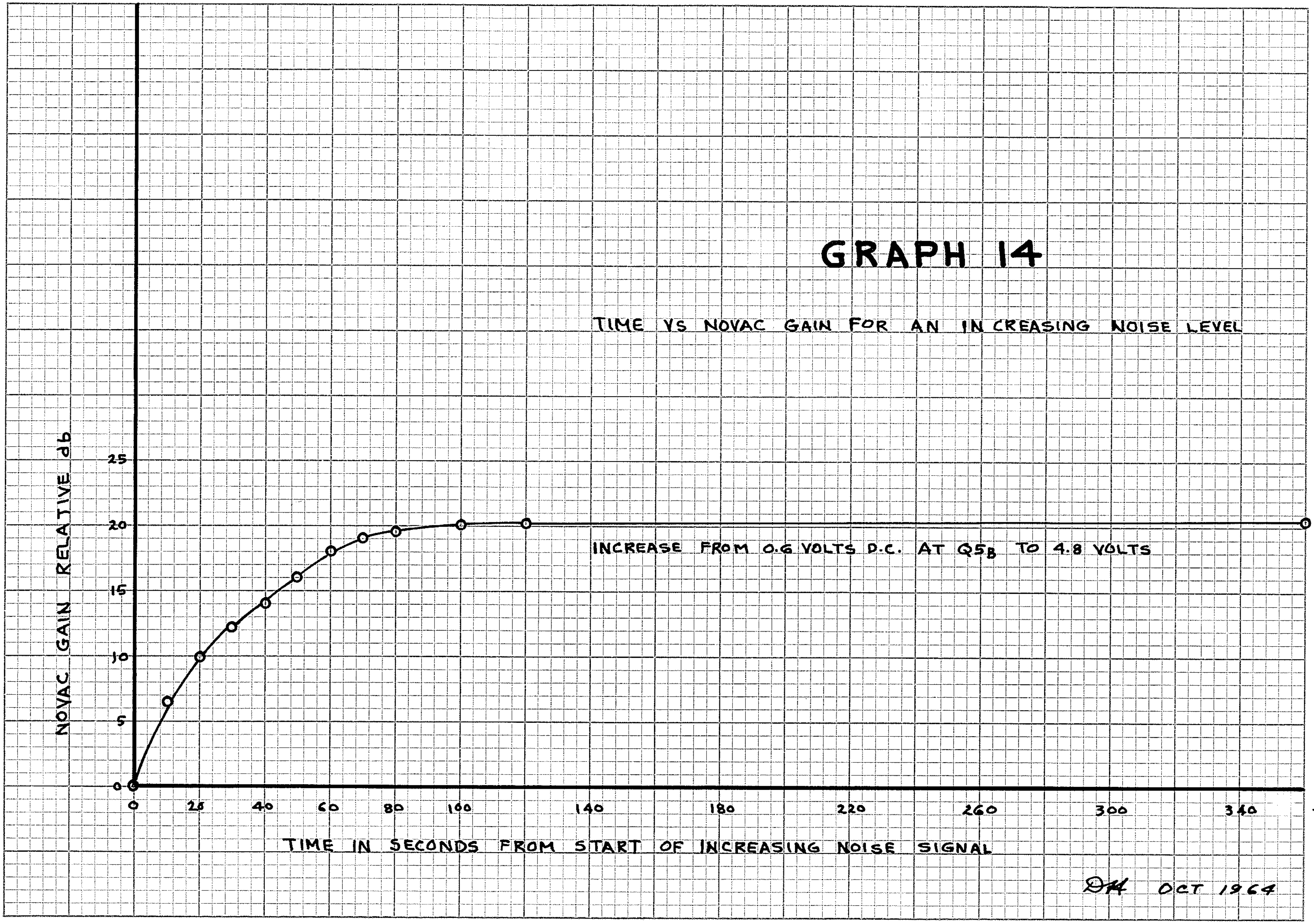
In order to obtain the results shown in Table 15, the background noise was suddenly increased from 50 db to 70 db above reference.

TABLE 15

TIME VS NOVAC GAIN FOR INCREASING NOISE LEVEL

Time (secs.)	$Q5_B$ Volts D.C.	NOVAC Gain (rel. db)
0	.6	0
10	1.0	6.5
20	1.4	10.0
30	1.8	12.0
40	2.3	14.0
50	2.9	16.0
60	3.5	18.0
70	3.9	19.0
80	4.2	19.5
90	4.35	19.8
100	4.45	20.0
110	4.55	20.1
120	4.65	20.2
360	4.80	20.5

These results were plotted on GRAPH 14.



GRAPH 14
TIME VS NOVAC GAIN FOR AN INCREASING NOISE LEVEL
INCREASE FROM 0.6 VOLTS D.C. AT Q5B TO 4.8 VOLTS
NOVAC GAIN RELATIVE db
0
5
10
15
20
25
0
20
40
60
80
100
140
180
220
260
300
340
TIME IN SECONDS FROM START OF INCREASING NOISE SIGNAL
DH OCT 1964

4-7-2 Time vs Gain for a Decreasing Noise Signal

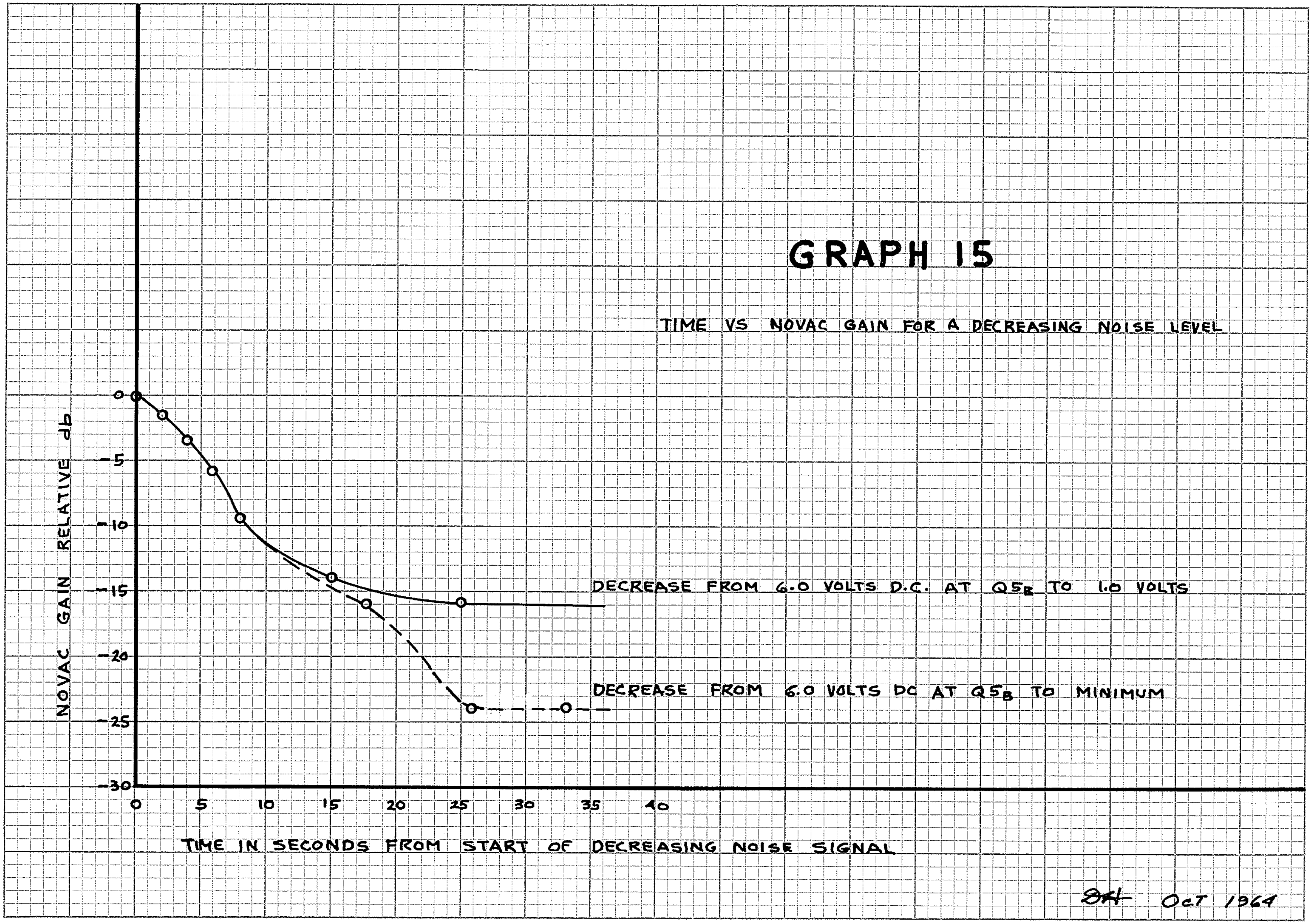
(1) In order to obtain the results shown in Table 16, the background noise was suddenly decreased from 73 db to 56 db above reference.

TABLE 16

TIME VS GAIN FOR A DECREASING NOISE SIGNAL

Time (secs.)	D.C. Volts at $Q5_B$	NOVAC Gain (rel. db)
0	6.0	0
2	5.0	-1.5
4	4.0	-3.5
6	3.0	-6.0
8	2.0	-9.5
15	1.2	-14.0
25	1.0	-16.0

These results were plotted on GRAPH 15.


GRAPH 15
TIME VS NOVAC GAIN FOR A DECREASING NOISE LEVEL
DECREASE FROM 6.0 VOLTS D.C. AT Q5B TO 1.0 VOLTS
DECREASE FROM 6.0 VOLTS DC AT Q5B TO MINIMUM
NOVAC GAIN RELATIVE db
0
-5
-10
-15
-20
-25
-30
0
5
10
15
20
25
30
35
40
TIME IN SECONDS FROM START OF DECREASING NOISE SIGNAL
Oct 1964

(2) In order to obtain the results shown in Table 17, the background noise level was suddenly decreased from 73 db to 48 db above reference.

TABLE 17

TIME VS GAIN FOR A DECREASING NOISE SIGNAL

Time (secs.)	D.C. Volts at $Q5_B$	NOVAC Gain (rel. db)
0	6	0
2	5	-1.5
4	4	-3.5
6	3	-6.0
10	2	-9.5
18	1	-16.0
26	.5	-24.0
33	.3	-24.0

These results were plotted on GRAPH 15.

4-8 FREQUENCY RESPONSE OF 2 1/4 INCH SPEAKER USED AS A MICROPHONE

A 2 1/4 inch speaker, model No. LPH 65, manufactured by Lorenz of Germany was used as the noise sensing microphone for the NOVAC device. In tests, it had been found that this device acted as a suitably sensitive noise sensing microphone for frequencies above 1,500 cps.

The frequency response was obtained as follows:

(1) A calibrated Bruel and Kjoer sound level meter was placed approximately 3 feet from a speaker. By means of an oscillator and amplifier, tones between 1,000 and 4,000 cps were reproduced by this speaker. The Bruel and Kjoer meter reading was recorded for each tone.

(2) The Bruel and Kjoer meter was removed and the Lorenz unit mounted in its place. The same tones as before were reproduced by the oscillator, amplifier and speaker. The open circuit voltage produced by the Lorenz unit was recorded for each tone.

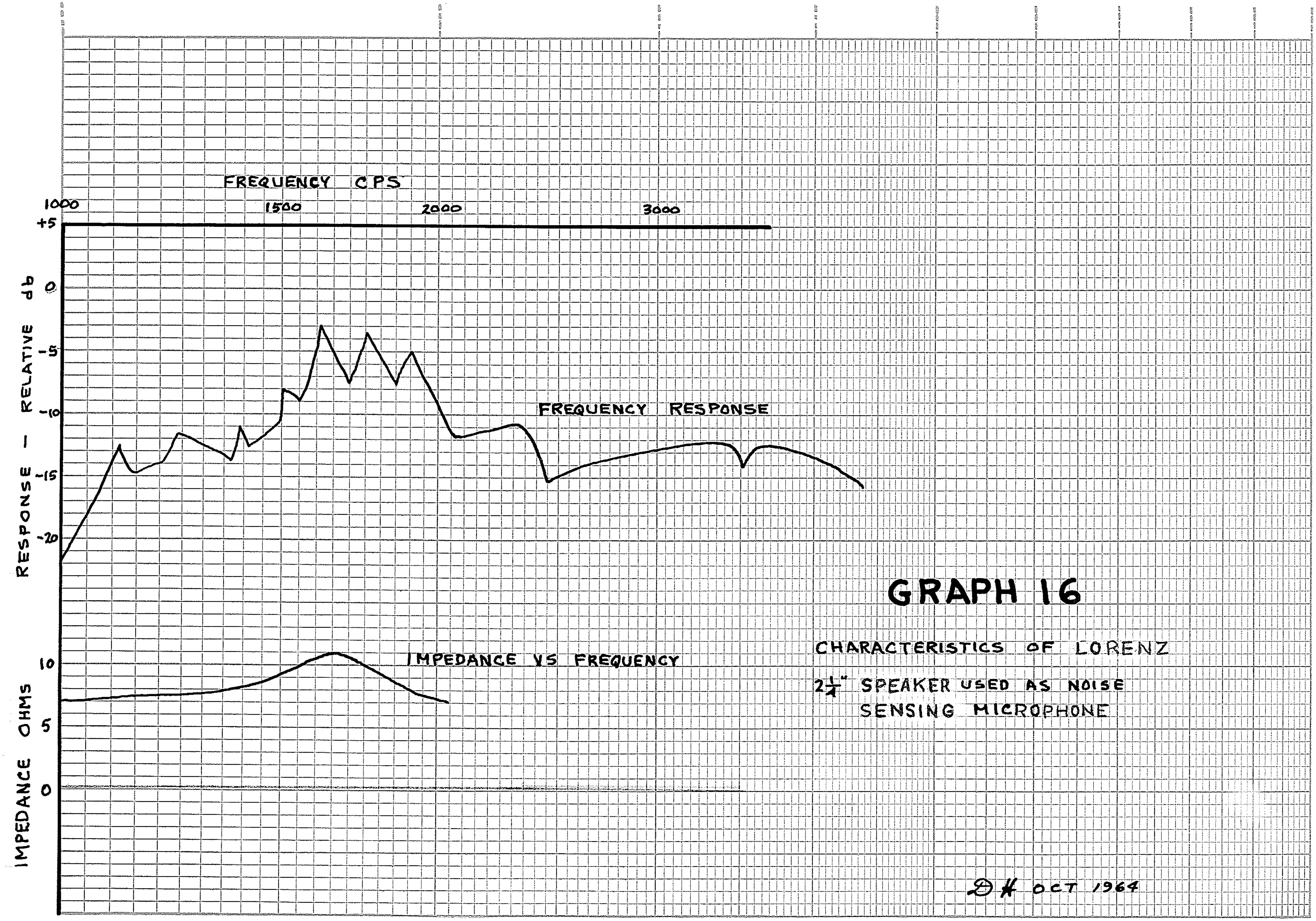
(3) The readings obtained by the Bruel and Kjoer meter and those obtained from the Lorenz unit were compared. These results were corrected by the Bruel and Kjoer calibration graph.

The corrected frequency response of the Lorenz speaker was plotted on GRAPH 16.

The physical appearance of the Lorenz speaker is shown in FIG. 8, page 42.

4-9 IMPEDANCE OF 2 1/4 INCH SPEAKER USED AS A MICROPHONE

The impedance of the 2 1/4 inch speaker was measured by standard techniques and plotted in GRAPH 16.


FREQUENCY CPS
1000
1500
2000
3000
+5
0
-5
-10
-15
-20
RESPONSE - RELATIVE db
FREQUENCY RESPONSE
10
5
0
IMPEDANCE OHMS
IMPEDANCE VS FREQUENCY
GRAPH 16
CHARACTERISTICS OF LORENZ
2¼" SPEAKER USED AS NOISE
SENSING MICROPHONE
OCT 1964

4-10 FREQUENCY RESPONSE OF NOVAC

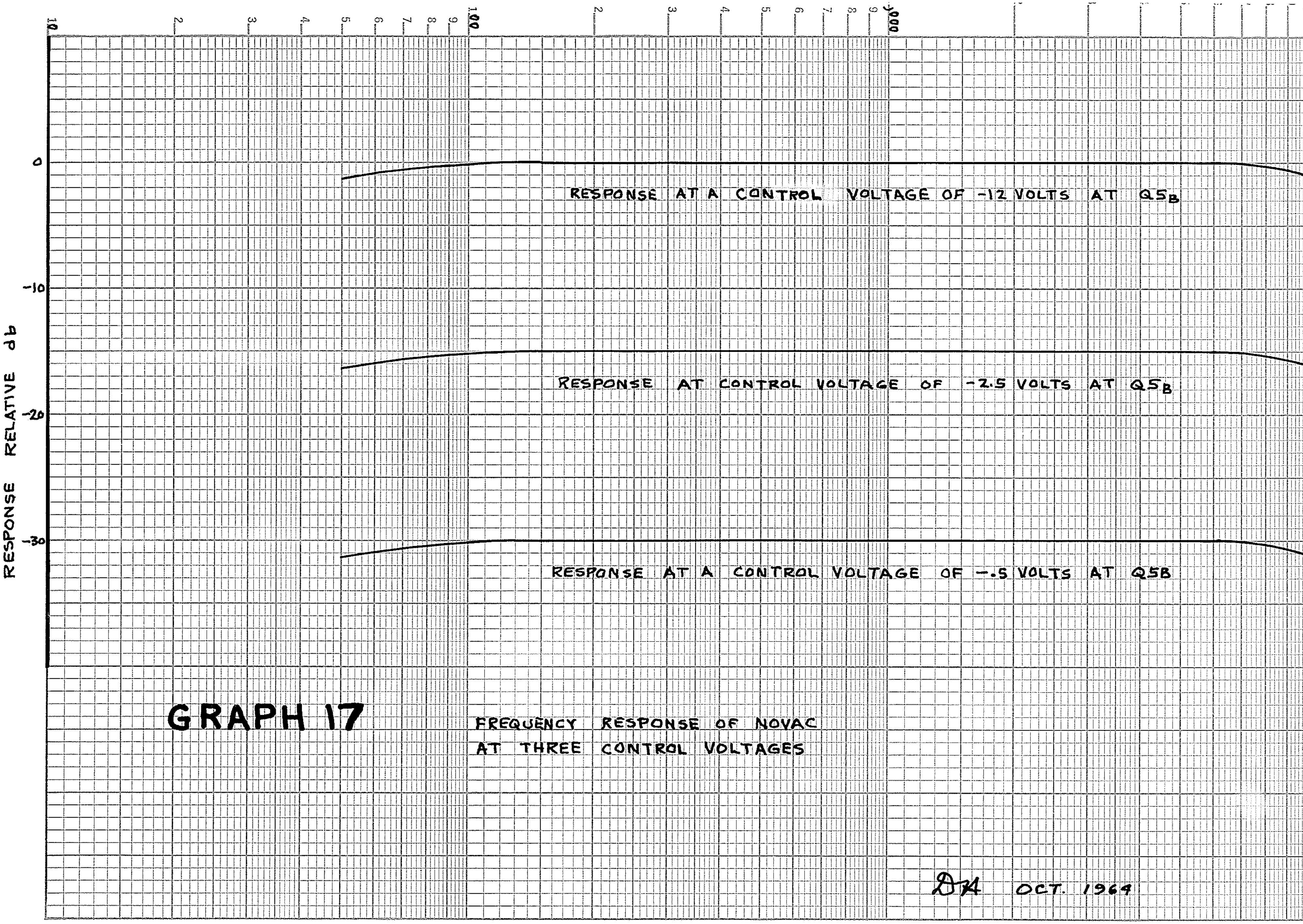
The frequency response of the NOVAC was measured by standard techniques and was plotted on GRAPH 17 for three levels of equivalent background noise.

4-11 TESTS IN ACTUAL PREMISES

The NOVAC device was installed in the Embassy Grill restaurant, and adjusted for a desirable music signal to noise ratio when the restaurant was about one quarter filled.

The three listeners that had given their opinions on previous matters were asked to monitor the music signal in relation to the background noise and to ascertain whether a suitable music signal to noise ratio was maintained as the restaurant became noisier and thence subsequently quieter. It was their opinion that the music level did follow the noise level satisfactorily. At times, when a large number of people left the restaurant at one time, and thus reduced the background noise rapidly, the music level was momentarily higher than desirable. However, for the short time that this occurred, it was not considered objectionable.

Large noises caused by trucks going by did not noticeably affect the level of the music, nor did sudden noises such as cash register noise. The NOVAC was judged to perform satisfactorily in this environment.



GRAPH 17

FREQUENCY RESPONSE OF NOVAC
AT THREE CONTROL VOLTAGES

At a later date, another NOVAC unit was placed in a Safeway store. Visits and listening tests were made when the store was busy and when it was idle. The ratio of music level to background noise was found satisfactory each time. At one time, after the installation had been in approximately three months, the manager of the store asked that the music level be checked. This was done and the level was found to be about 3 db higher than normal. This could have been caused by drift in the gain of the amplifiers used due to aging of components. With this one exception, the unit was considered to have operated satisfactorily.

CHAPTER FIVE

DISCUSSION

In Chapter Two "Defining of Design Objectives" some interesting, if somewhat limited, empirical results were obtained. It was found that, for a small group of male experts, the desirable music signal to background noise ratio appeared to be independent of the background noise level. This observation was found to exist over background noise levels between 50 and 80 db above reference. The music was considered best reproduced approximately 1 db lower than the background noise. Possible reasons for the negative signal to noise ratio being desirable were thought to be:

(1) In making their level adjustment, the background music was normally kept as low in level as possible by the experts since it was considered not desirable for the music to be annoyingly loud.

(2) The background noise level varied from second to second, whereas the background music, being purposely restricted to a narrow dynamic range, did not. Thus, even at a negative signal to noise ratio, the background music tended to fill in where the noise level was lower than the music signal.

It was the opinion of these same people that, at a background noise level of 72 db above reference, the music level could vary ± 4 db from the desired level and still be tolerable. Beyond this ± 4 db range, the music level was considered to have been too low or too high in level.

In some stores and restaurants, background noise levels were found to vary as much as ± 10 db. This tended to show the need for a NOVAC type device.

In Chapter Three the following observations were made:

(1) After conducting listening tests, it was thought that a low frequency (120 to 190 cps) noise sensing band was unsatisfactory due mainly to large low frequency noises such as those caused by trucks, etc., that were not representative of the noise as heard by a normal listener and yet caused large and annoying changes in the gain of the NOVAC type device, and subsequently the music level.

(2) After conducting further listening tests, it was also considered that the noise spectrum above 6 K.C. was unsatisfactory for sensing the background noise. It was considered that an omnidirectional noise sensing microphone would be necessary to sense noise from all directions. Microphones with this characteristic for this frequency

range were considered too expensive for this practical project. Also, it was considered that, due to the character of the noise above 6 K.C. being highly impulsive in character and on the average weak in strength, it would be overly costly to use this signal for controlling the music level.

(3) In the third and final attempt, the background noise was sensed in a slot located in the middle of the noise spectrum and was found to give satisfactory results.

A few problems were still encountered with this method. It was found that the first constructed slot rejection filter adversely affected the quality of the music. This slot filter effectively rejected all frequencies between 1,150 and 1,650 cps from the music signal. However, after subjective A, B tests it was considered that a slot filter that rejected the band of frequencies from 1,540 to 1,780 cps in the music signal did not cause a serious change in the quality of the music. Also, it was found that the slot rejection and bandpass filters for this band of frequencies could simply and economically be constructed. As these filters were found satisfactory from a technical and economic point of view, no further tests were made. However, it is felt that the effect of slots in the music signal bears further investigation.

Another problem was that of sudden loud noises, which caused undue changes in the NOVAC gain. This problem was much reduced by the development of the "10 db peak to rms noise clipper" that limited sudden noise levels to no more than 10 db above the average noise level. The figure of 10 db was considered to be suitable after testing on a trial and error basis.

After further experimenting on a trial and error basis, it was considered that the D.C. control time constant, FIG. 1, page 22, of the variable loss device should be approximately one minute. This prevented the music from attempting to follow minor variations in the noise signal and yet allowed the music signal to increase at a rapid enough rate to follow normal increases in the noise level. However, rapid decreases in noise due, for instance, to people leaving a restaurant, were considered to be a problem in that the time constant of the control R.C. filter was too long. This caused the music level to stay at a higher than desirable level for a noticeable period of time after the decrease in noise. In order to overcome this, a diode, D1, was introduced (FIG. 1, page 22) to decrease the decay time but at the same time not affect the desirable long charging time constant.

After evaluating the problems and the empirical results, a prototype unit was constructed and tested. The following is a discussion of the tests conducted on the completed NOVAC prototype:

(1) NOVAC Gain vs Background Noise

The curve, shown in GRAPH 9, page 57, showed a good linearity between the background noise signal and the NOVAC gain. The deviation in linearity between the noise level and the NOVAC gain between the noise levels of 50 and 80 db above reference was less than 1 db, which was better than what was considered necessary.

This linearity was shown in GRAPH 10, page 59, as well, which was obtained by using a different selection of noise for each of the recorded measurements. The straight line drawn through these points indicated a one to one relationship between the noise level and the NOVAC gain except for the portion of the curve when the noise level was greater than 80 db above reference. Since 80 db above reference was considered the maximum design noise level, this departure from linearity was considered satisfactory.

In GRAPH 11, page 61, the effect of a 1,680 cps tone (used instead of background noise) was shown on the NOVAC gain. The test was similar to test #1 which was conducted using background noise as the control signal.

However, instead of a continuously varying noise signal, the control signal in this case was a steady tone of 1,680 cps. As can be seen on GRAPH 11, page 61, the one to one relationship was quite linear, except at the lower tone levels, where the gain of the NOVAC dropped less than for a corresponding drop in the tone level. However, it was thought that the tone did not accurately represent background noise. For instance, background noise had random variations in level with time whereas the tone did not. Thus it was thought that the more indicative measurement was one that used actual background noise. As previously discussed, the relationship between actual background noise and the NOVAC gain was found to be reasonably linear on a one to one basis, from 50 to 80 db of noise above reference.

(2) Effect of Music Level on NOVAC Gain

As can be seen by the calculated results in Table 9, page 68, the music playing through the NOVAC did have an effect on the gain of the NOVAC. This was because the slot rejection and bandpass filters were not ideal and some music signal did bypass them and thence combined with the background noise to increase the NOVAC gain. By measurement and subsequently calculation, it was found that a music signal to noise ratio of 12 db caused, on the average, a 2.5 db increase in NOVAC gain. This was the maximum increase that was considered acceptable. As the music signal to noise ratio was increased, the music level effect on the NOVAC gain became more and more pronounced until at a music signal to noise ratio of between 17 and 22 db, the NOVAC gain went into a runaway condition and the NOVAC gain increased to the maximum possible. A single accordion selection was found to produce a greater change in the NOVAC gain than the other selections. This was not considered serious for practical purposes because:

(i) This type of selection was found to be seldom played.

(ii) In order to help overcome this problem in a practical installation, the noise sensing microphone could be placed in the same plane as the speakers used for reproducing the music. This would tend to reduce the coupling from the music speaker to the noise sensing microphone due to their directional characteristics.

However, it was thought that this problem must be considered in future practical tests and applications to ensure that it was not a serious problem.

From Table 9, page 68, which was derived by using music signals for testing the rejection of the slot rejection and bandpass filters to the music signal, it was calculated that the music signal was rejected from entering the noise bandpass filter by 16 db. In the steady state testing of the slot rejection and bandpass filters shown in GRAPH 8, page 51, and described in Chapter Two, page 8, it was calculated that the music signal was rejected from entering the noise bandpass filter by 21 db.

The difference between 16 db and 21 db was difficult to explain. One reason for this difference may have been that in the derivation of Table 9, page 68, the music signal rejection was measured by averaging a fluctuating signal, with subsequent errors. Also, the 21 db rejection figure was that determined by the slot rejection and bandpass filters only. This figure could have been changed by the frequency response of the speakers used for reproducing the music and also by the frequency response of the room itself. If the music speaker or the room had happened to have a peak in its response in the vicinity of the slot, this could have decreased the effective music rejection. Also, distortion products of the music power amplifier could have entered the noise bandpass filter and thus reduced the effective rejection of the music signal.

It was thought that if a larger music signal to noise ratio was required in future applications, that the slot rejection and bandpass filters could be improved with subsequent higher costs. This was not attempted with this unit since it was believed that the filters were satisfactory in their present form.

(3) Distortion of NOVAC to Music Signals

It was found in these tests, page 72, that if the peak music input signal to the NOVAC was kept below 2 mv, the distortion of the output signal was always less than two percent, regardless of the D.C. control voltage at $Q5_B$. It was considered that this figure of two percent was acceptable and thus no problem due to excessive distortion was encountered.

(4) Charge and Discharge Times of Control Circuit of NOVAC

As can be seen on GRAPH 14, page 76, the initial rate of rise for an increasing background noise was .65 db per second. Approximately 50 seconds was required before the NOVAC gain increased to 80 percent of its final value.

On the other hand, for a decreasing noise signal, the initial rate of decrease, as shown on GRAPH 15, page 78, was .75 db/second and approximately 10 seconds was required for the NOVAC gain to drop to 80 percent of its final value for curve (a). For curve (b) the time required was about 20 seconds.

It was considered that in normal use the relatively slow rise time helped to prevent sudden loud noises from unduly affecting the NOVAC gain and the relatively fast decay time helped the NOVAC gain to follow a normal drop in background noise.

(5) Frequency Response and Impedance of 2 1/4 Inch Lorenz Speaker Used as a Noise Sensing Microphone

As shown on GRAPH 16, page 81, both the impedance and frequency response of the speaker used as a microphone peaked in the area of 1,650 cps, which was the noise sensing band. This tended to help the electrical characteristics of the noise bandpass filter in rejecting all but the designated band of noise frequencies.

(6) Frequency Response of NOVAC to Music Signals

As shown in GRAPH 17, page 83, the frequency response of NOVAC to the music signal was flat within ± 3 db from 50 to 10,000 cps and was thus considered adequate.

(7) Tests in Actual Premises

These tests showed that, subjectively, the prototype NOVAC performed satisfactorily, following the background noise level with corresponding changes in music level. The test undertaken at the Safeway store indicated that the particular unit installed at that location had caused the music level to increase approximately 3 db above that considered desirable. This occurred over a three month interval of time. It was thought that this relatively long period of time between adjustments would not be considered unsatisfactory.

The following is a list of unanswered questions that occurred to the author but which required research beyond the limits of this undertaking:

(1) What is the preferable background music level to noise level ratio in all types of establishments and for all types of noise?

(2) What is the range above and below the preferable level over which the ratio can still be considered satisfactory by large numbers of observers?

(3) What is the effect on the character of the music signal when different portions of the frequency spectrum are removed by means of slot rejection filters?

CHAPTER SIX

CONCLUSION

The practical objective of this thesis was to build a NOVAC device that would automatically control a background music signal in the presence of a varying background noise level so as to produce a satisfactory music signal to noise ratio.

One method* of obtaining this objective has been examined and a prototype device built. The device was found satisfactory but having certain restrictions.

These restrictions were:

(1) The background noise character should be constant with time and only the noise level should vary.

(2) The maximum music signal to background noise should be 12 db (for this design).

(3) The background noise level should be between 50 and 80 db above reference (for this design).

The particular design arrived at was based on meeting the minimum design requirements with minimum costs. It is estimated that the cost of the parts for this particular design was less than $45.00.

* See page 5.

If it was desired to improve the NOVAC device as designed and evaluated in this thesis, it is thought that costs would have to be increased substantially. Keeping this in mind, the following are possible improvements that could be made and that bear further investigation:

(1) The slot rejection and bandpass filters could be made much narrower and sharper. This would help to prevent the music quality from being adversely affected and also would help to prevent the music signal from entering the noise sensing channel with its subsequent feedback problem.

(2) More than one noise sensing slot could be used. This would help in making the noise sensing portion of the NOVAC device more representative of different kinds of noise; i.e., noises having different spectra.

(3) A better noise clipper could be made that would more adequately suppress the effect of sudden unrepresentative noises.

These areas of investigation were not further studied since it was felt that the unit as it was originally designed was economically and technically satisfactory.

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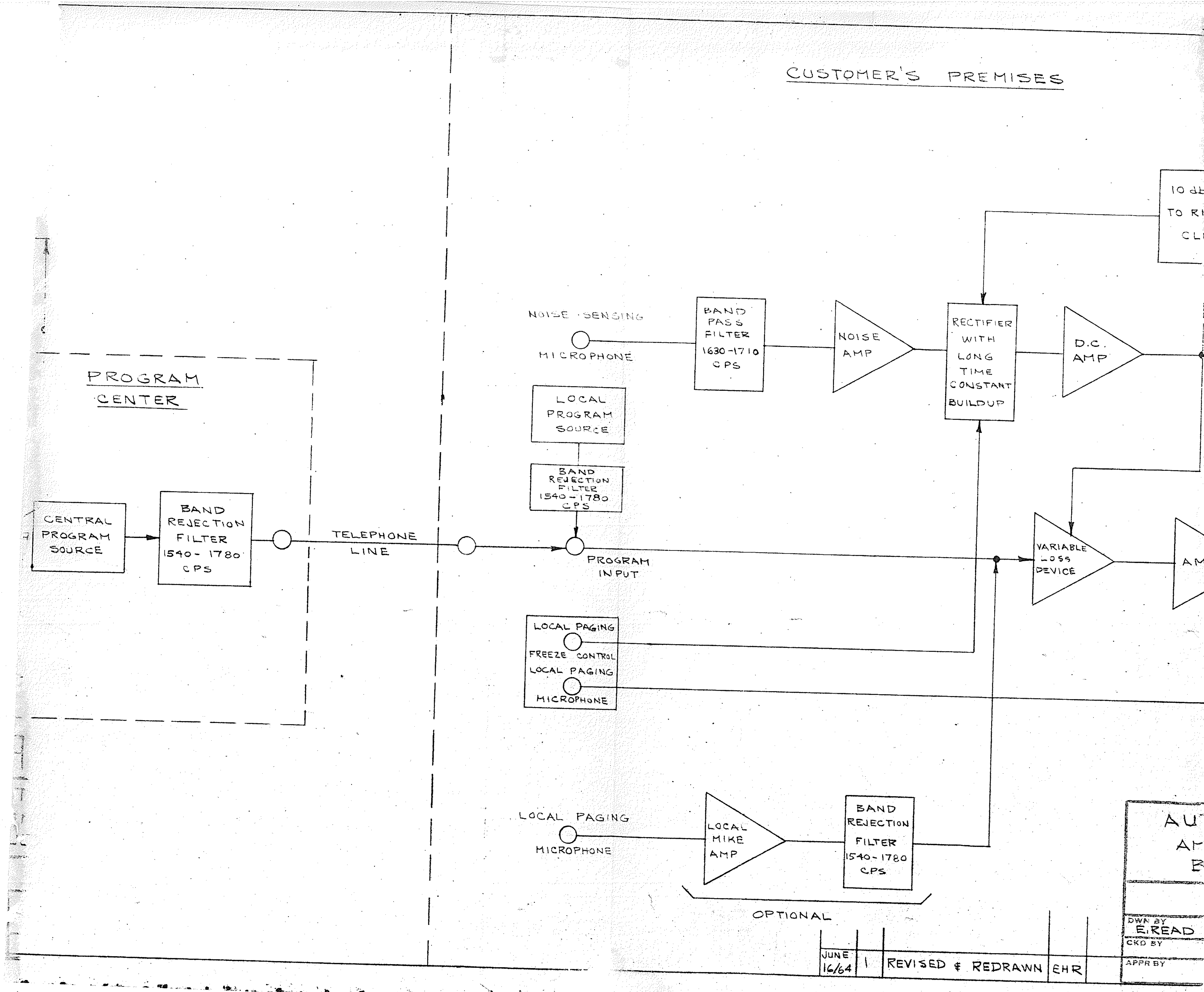
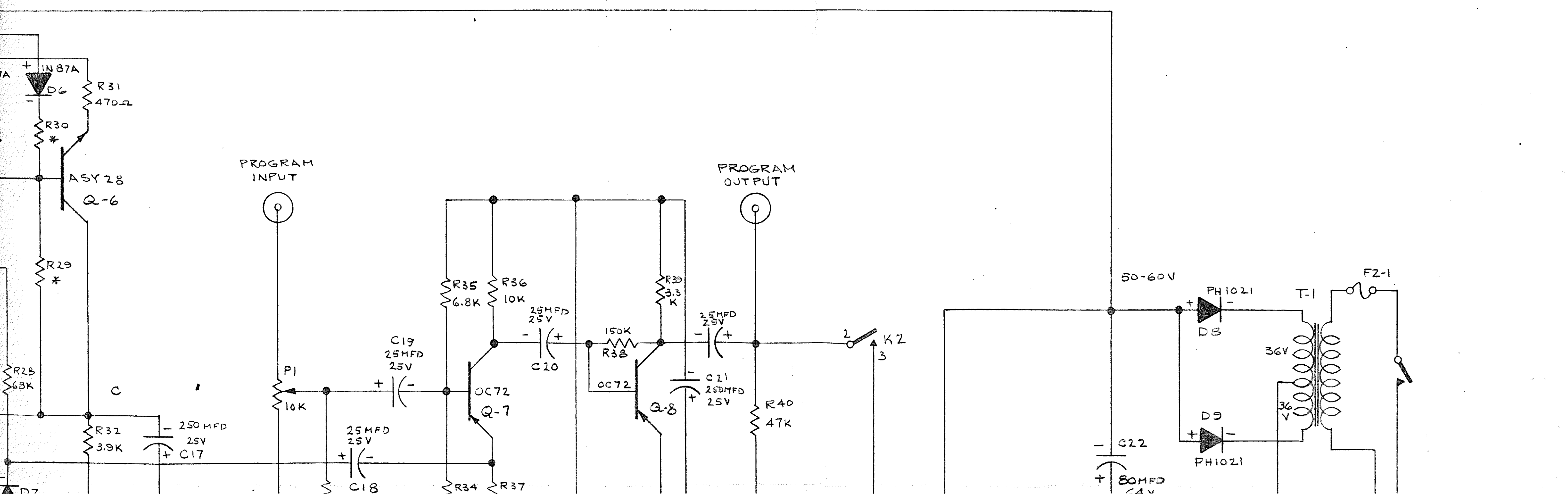
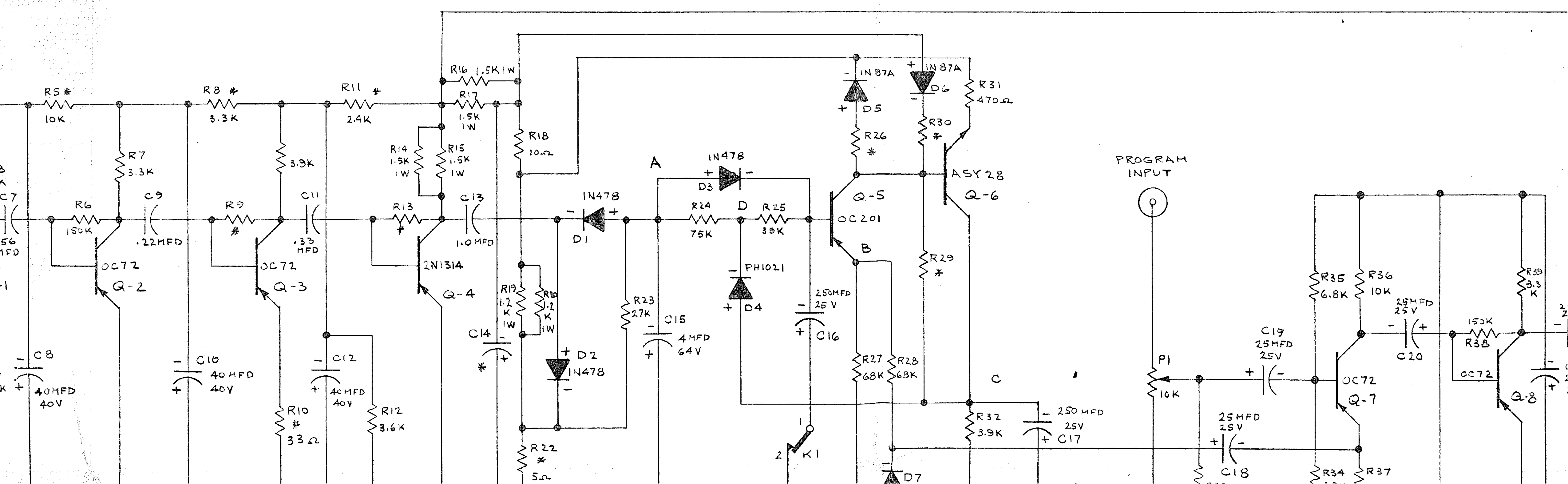
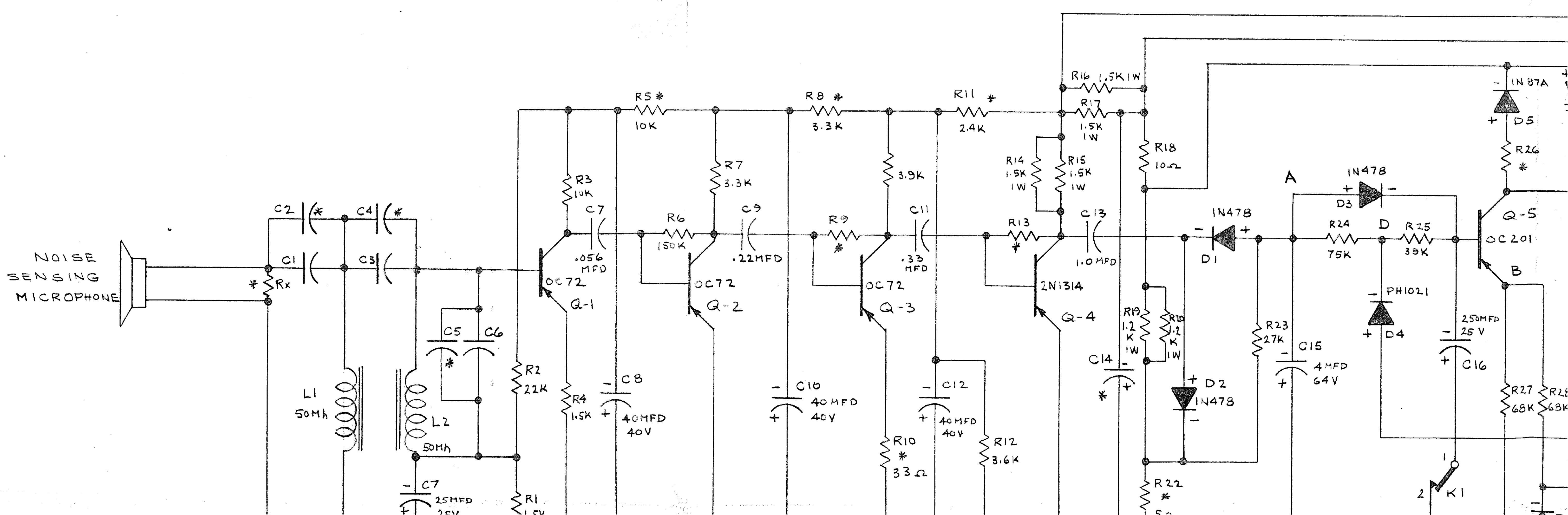

CUSTOMER'S PREMISES
PROGRAM CENTER
CENTRAL PROGRAM SOURCE
BAND REJECTION FILTER 1540- 1780 CPS
TELEPHONE LINE
NOISE SENSING MICROPHONE
BAND PASS FILTER 1630-1710 CPS
NOISE AMP
RECTIFIER WITH LONG TIME CONSTANT BUILDUP
D.C. AMP
10 db
TO RM
CLIP
LOCAL PROGRAM SOURCE
BAND REJECTION FILTER 1540 - 1780 CPS
PROGRAM INPUT
VARIABLE LOSS DEVICE
AM
LOCAL PAGING
FREEZE CONTROL
LOCAL PAGING MICROPHONE
LOCAL PAGING MICROPHONE
LOCAL MIKE AMP
BAND REJECTION FILTER 1540-1780 CPS
OPTIONAL
AU
AM
B
DWN BY E.READ
CKD BY
APPR BY
JUNE 16/64
1
REVISED & REDRAWN
EHR


IN 87A
D6
R31
470Ω
R30
*
ASY 28
Q-6
R29
*
R28
68K
C
R32
3.9K
250 MFD
25V
C17
D7
PROGRAM
INPUT
P1
10K
C19
25MFD
25V
25MFD
25V
C18
R35
6.8K
R36
10K
25MFD
25V
C20
OC72
Q-7
R34
R37
150K
R38
R39
3.3
K
OC72
Q-8
25MFD
25V
C21
250MFD
25V
PROGRAM
OUTPUT
R40
47K
K2
50-60V
PH1021
D8
T-1
FZ-1
36V
36
V
D9
PH1021
C22
80MFD


R5 *
10K
R6
150K
R7
3.3K
C7
C8
40MFD
40V
C9
.22MFD
OC72
Q-2
R8 *
3.3K
R9
*
3.9K
C10
40MFD
40V
C11
.33
MFD
OC72
Q-3
R10
*
33Ω
R11 *
2.4K
C12
40MFD
40V
R12
3.6K
R13
*
2N1314
Q-4
R14
1.5K
1W
R15
1.5K
1W
R16 1.5K 1W
R17
1.5K
1W
C13
1.0MFD
C14
*
R18
10Ω
R19
1.2
K
1W
R20
1.2
K
1W
R22
*
5Ω
1N478
D1
D2
1N478
R23
27K
C15
4MFD
64V
A
1N478
D3
R24
75K
D
R25
39K
PH1021
D4
Q-5
OC201
B
250MFD
25V
C16
1N87A
D5
R26
*
1N87A
D6
R30
*
R31
470Ω
ASY28
Q-6
R29
*
R27
68K
R28
68K
C
R32
3.9K
250MFD
25V
C17
1
2
K1
D7
PROGRAM
INPUT
P1
10K
C19
25MFD
25V
25MFD
25V
C18
R35
6.8K
R36
10K
25MFD
25V
C20
OC72
Q-7
R34
R37
150K
R38
R39
3.3
K
OC72
Q-8



NOISE SENSING MICROPHONE
Rx
C1
C2
C3
C4
C5
C6
C7 25MFD
L1 50Mh
L2 50Mh
R1
R2 22K
R3 10K
R4 1.5K
C7 .056 MFD
C8 40MFD 40V
OC72 Q-1
R5 10K
R6 150K
R7 3.3K
C9 .22MFD
OC72 Q-2
C10 40MFD 40V
R8 3.3K
R9
3.9K
C11 .33 MFD
OC72 Q-3
R10 33Ω
C12 40MFD 40V
R11 2.4K
R12 3.6K
R13
R14 1.5K 1W
R15 1.5K 1W
R16 1.5K 1W
R17 1.5K 1W
C13 1.0MFD
2N1314
Q-4
C14
R18 10Ω
R19 1.2 K 1W
R20 1.2 K 1W
1N478 D1
D2 1N478
R22
R23 27K
C15 4MFD 64V
A
1N478 D3
R24 75K
D
R25 39K
PH1021 D4
C16 250MFD 25V
K1
1N87A D5
R26
Q-5 OC201
B
R27 68K
R28 68K

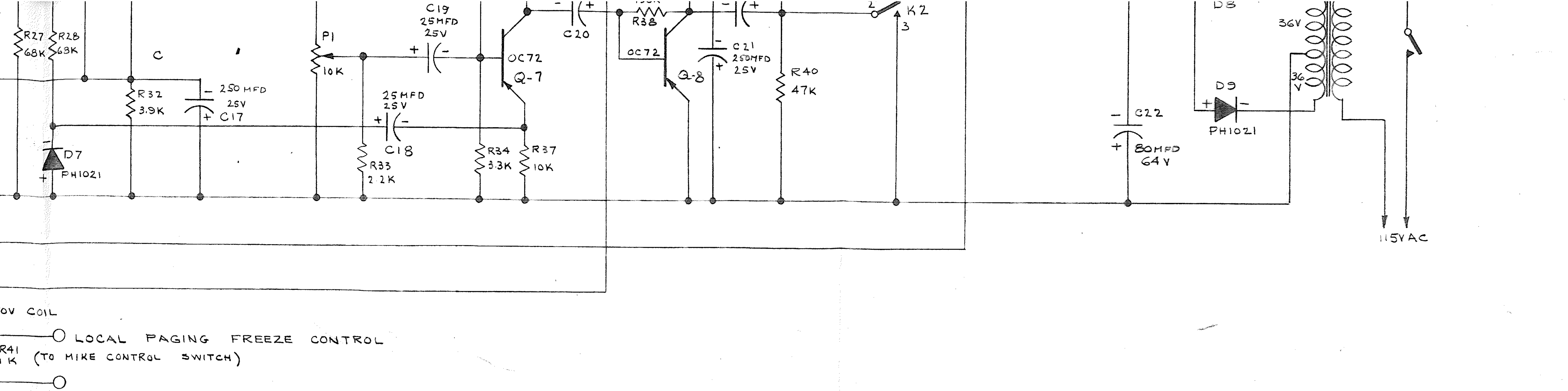
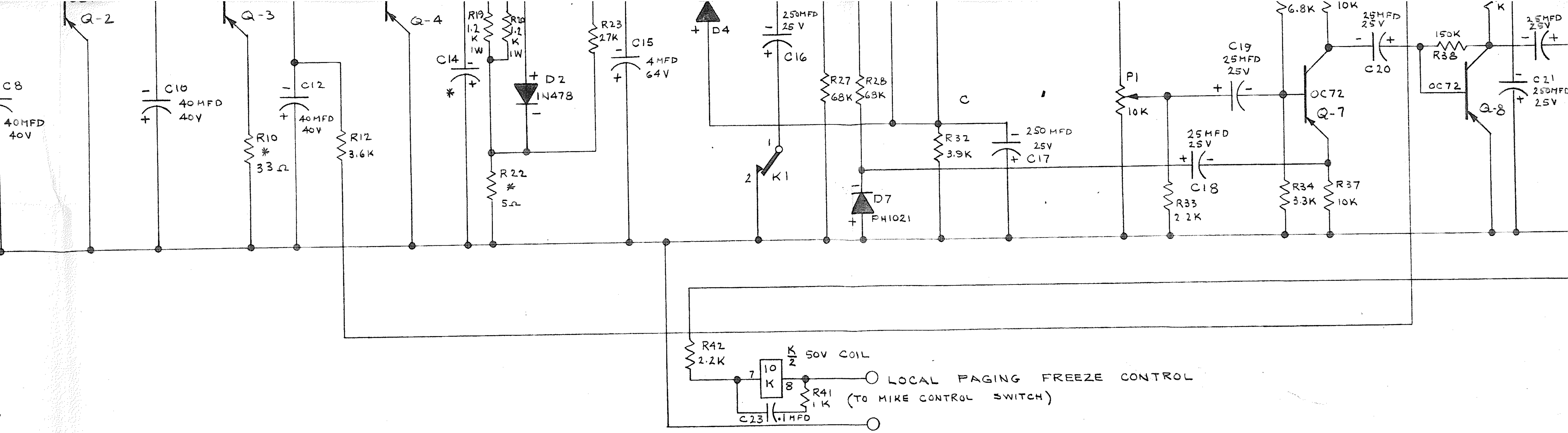
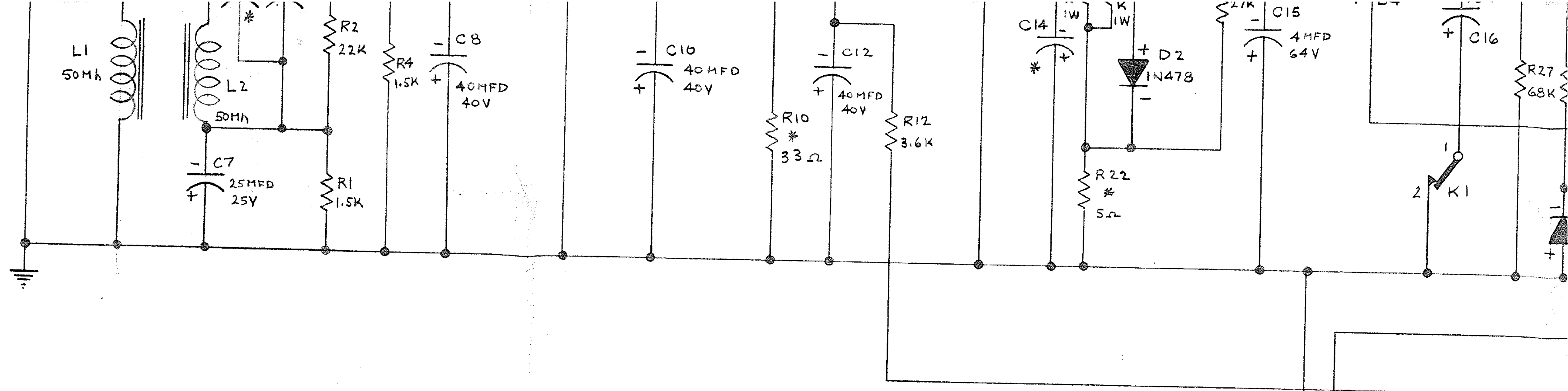
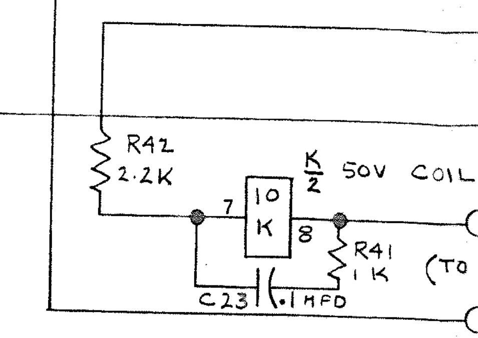


FIG. 7

AUTOMATIC VOLUME AMPLIFIER (NOVAC)
SCHEMATIC DIAGRAM


Q-2
Q-3
Q-4
C8
40MFD
40V
C10
40MFD
40V
C12
40MFD
40V
R10
33Ω
R12
3.6K
R19
1.2K
1W
C14
D2
1N478
R22
5Ω
R23
27K
C15
4MFD
64V
D4
250MFD
25V
C16
K1
R27
68K
R28
68K
D7
PH1021
R32
3.9K
250MFD
25V
C17
P1
10K
C19
25MFD
25V
25MFD
25V
C18
R33
2.2K
6.8K
10K
OC72
Q-7
R34
3.3K
R37
10K
25MFD
25V
C20
150K
R38
OC72
Q-8
C21
250MFD
25V
R42
2.2K
K/2 50V COIL
R41
1K
C23
.1MFD
LOCAL PAGING FREEZE CONTROL
(TO MIKE CONTROL SWITCH)





* - SELECTED RESISTERS & CAPACITORS
ALL RESISTERS ½W UNLESS OTHERWISE SPECIFIED.
ALL CAPACITORS 125V UNLESS OTHERWISE SPECIFIED.

FIG. 7

AUTOMATIC VOLUME
AMPLIFIER (NOVAC)
SCHEMATIC DIAGRAM

RIMAC LTD
WINNIPEG MAN

JUNE 16/64	1	NOTES DELETED	EHR

DWN BY E. READ	DATE MAY 28/64	SCALE NONE
CKD BY	DATE	DWG NO 151D
APPR BY	DATE	