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Sound reproduction in the field of music : a survey and analysis

Arthur John Holton
University of the Pacific

SOUND REPRODUCTION IN THE FIELD OF MUSIC
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Arthur John Holton

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FOREWORD

Music reproduction through electronic instruments is rapidly becoming a highly complex and somewhat controversial field of study and development. The technicians and designers are constantly adding new recording devices and reproducers with an endless variety of models with internal and external modifications to an already confusing array of commercial offerings. The critical listener who desires a good music reproducer must face this bewildering situation not only with a limited technical knowledge but with a lifetime of having heard only the average commercial radio phonograph with its many mechanical and physical limitations. This conditioning is so subtle and complete that many professional musicians actually prefer the limited presence and fidelity of an inadequate reproducer to the crystal clear "live" feeling that a so-called "high-fidelity" reproducer gives to music.[1]

This writer is not concerned with belaboring the issues surrounding these human failings. Rather, this thesis is an attempt to make use of as much technical knowledge as seems pertinent to a musician attempting to reproduce music with the utmost of realism and faithfulness to the original

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[1] See Chapter I, p. 13.

sound. The experiments and research that have gone into this thesis cover the years between 1940 and 1955, a period involving a world war and its many technical advancements, including the now commercially practical wire and tape recorders and the first laboratory development of a completely electronic recorder which has no moving parts[2] and uses a film recording that can be reproduced as easily as a large photograph.

In confronting such rapid technological development, the writer is convinced that the only lasting benefits this thesis might possess are toward furthering the education and conception of the musician and the technician in some of the physical, technical, musical, and psychological problems which must be faced before reproduced music can be truly satisfying to the critical listener.

The fact is that electronic technicians and engineers know too little about music and the musician knows too little about the science of electronics. It is this writer's opinion that each must study the other's field. The individuals who can achieve a fusion of the science of music reproduction and the art of music will approach the problems of dealing with musical sound with technical facility and musical sensitivity. [It is hoped that through this thesis the reader

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2 John Potter Shields, "New Sound Recording System," _Radio-Electronics_, 23:28, April, 1952.

will be stimulated to pursue the subject still further.] The possibilities in the future are fascinating to consider,[3] and there seem to be unlimited opportunities open to the enterprising individual in radio, television, the film industries, and the home.

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[3] Leopold Stokowski, Music for All of Us (New York: Simon and Schuster, 1943), pp. 228, 229, 256-60.

CHAPTER I

THE PROBLEMS THAT MUST BE FACED AND AN EXPLANATION OF TERMS THAT NEED TO BE UNDERSTOOD BY THE READER

When an individual spends many years studying and experimenting in any specialized field, it becomes increasingly difficult for that individual to communicate ideas and conclusions to the average reader without first establishing rapport through mutually understandable technical language and the recognition of the problems common to the field under observation and study. This thesis brings together the technical language and associated problems in the fields of electronics, acoustics, and music. Study in any one of these fields could constitute a life's work, and yet the purposes of this thesis demand a working knowledge in each field and a fusion of ideas and facts from all three to achieve its purpose.

The music student and musician seeking the adequate and artistic reproduction of music through electronic instruments constitutes a small minority group today. Chapter III of this thesis contains information showing that through the conditioning of present day radio, television, and commercially-made reproducers, many highly-trained musicians prefer medium or poor fidelity to the presence and "life" of high-fidelity sound. A second minority group consists of the radio and electronics minded "Hi-Fi bugs" who seek high-fidelity sound as a hobby, caring not at all for the artistic implications involved in reproducing music, but rather concentrating on the number of decibels and cycles per second being reproduced.

At least some of this first minority who are seeking better music reproduction have influenced a few reforms in commercial records and radio, but at this writing, artistic standards are generally very low and inconsistent. The trained musician and the electronics engineer can find very little common ground for communication when subjects such as harmonic balance, timbre, and dynamic expression in music reproduction are discussed. For the purpose of simplicity and clarity, we will label the musician seeking better music reproduction the "critical listener." The layman or casual listener, we will call the "average listener."[1]

The essential difference between the critical and average listener is found in the range of frequencies each can perceive under various conditions. The degree of loudness and the spectral[2] composition of the sound is one of

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[1] Jensen Radio Manufacturing Company, Technical Monograph Number Three, "Frequency Range and Power Considerations in Music Reproduction," (Chicago: Jensen Radio Manufacturing Company, 1944), p. 4.

[2] Spectral refers to the number and mixture of various pitchs that are part of the total sound.

these conditions, and the other is the level and character of the ambient noise background in which the listener is immersed.[3] This last condition is exceedingly important. Since we are constantly exposed to a never-ending background of noise that varies in loudness and quality from moment to moment, we must always keep in mind that noise acts as though it deafens the individual situated within it. This means that even the most perceptive ear cannot hear musical sounds, no matter how perfectly they are produced, if the level of the musical sound falls below that of the surrounding noise level.

In their pamphlet "Frequency Range and Power Considerations in Music Reproduction," the Jensen Company cites D. F. Seacord, D. F. Hoth, Harvey Fletcher and W. A. Munson while giving the following information:

> Seacord has published the results of about 2,200 measurements of room noise which indicate that the annual average residential noise level is 43 db. Only 5% of the residences had a noise level of 33 db or less, which checks closely with previously reported measurements in very quiet residences. This gives us two significant room noise levels for which the corresponding masking levels may be obtained by assigning typical spectral distribution to the noise then calculating the masking contours from these spectra. These contours are shown in Fig. 3.
>
> We can now combine the normal hearing contours with the masking contours in a variety of ways, if we choose, in order to determine the ability of a statistical

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[3] Jensen, loc. cit.

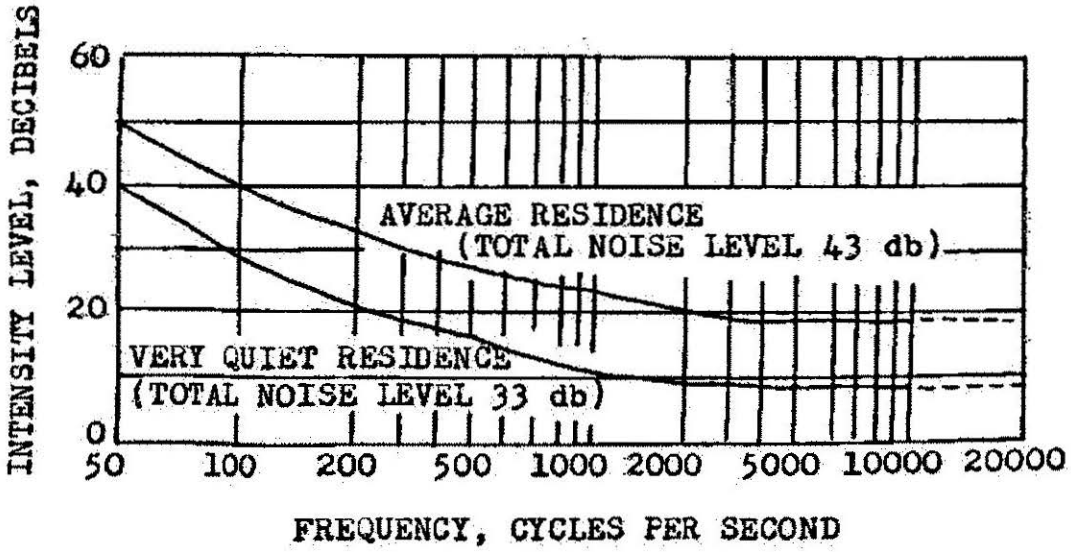

FIGURE 1*

MASKING LEVEL CONTOURS FOR NOISE IN AVERAGE

AND VERY QUIET RESIDENCES

(JENSEN, "FIGURE 3")

* Jensen, p. 4.

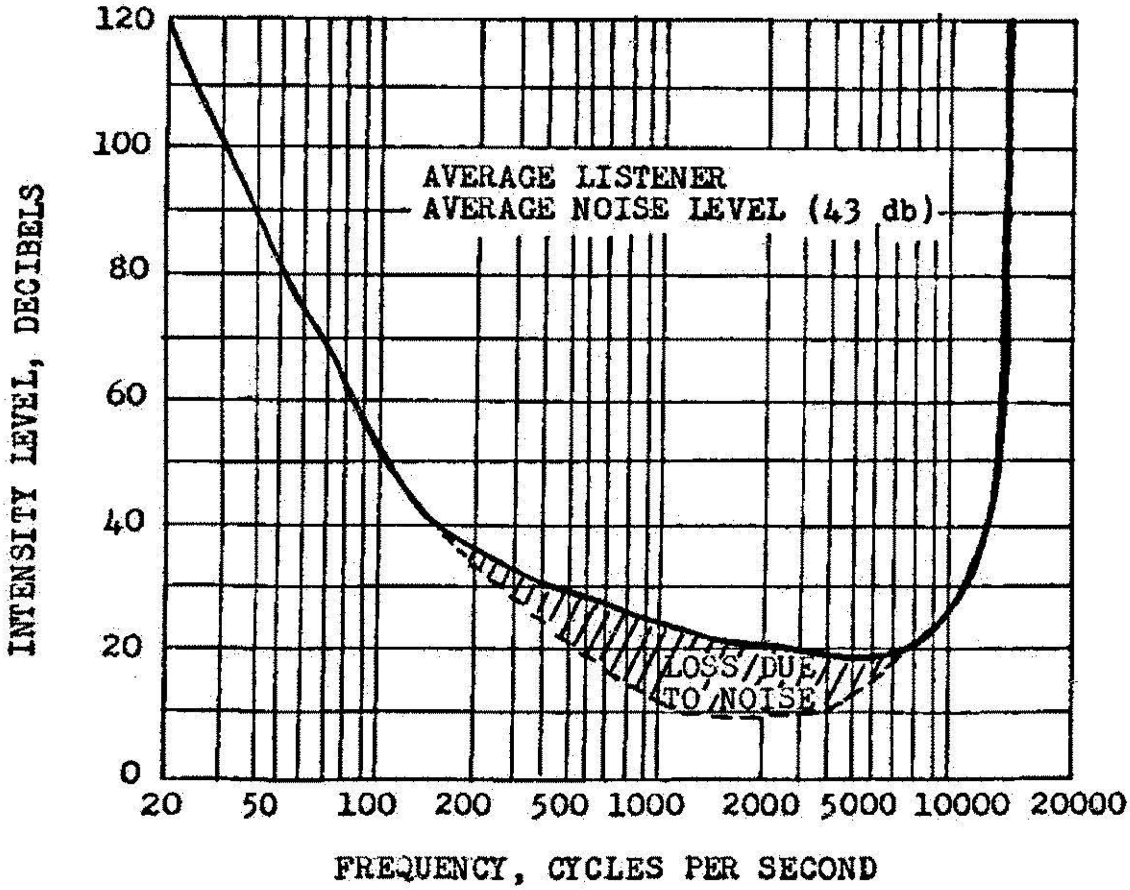

FIGURE 2*

EFFECTIVE HEARING CONTOUR FOR AVERAGE LISTENER

SITUATED IN AVERAGE NOISE

(JENSEN, "FIGURE 4")

* Jensen, p. 5.

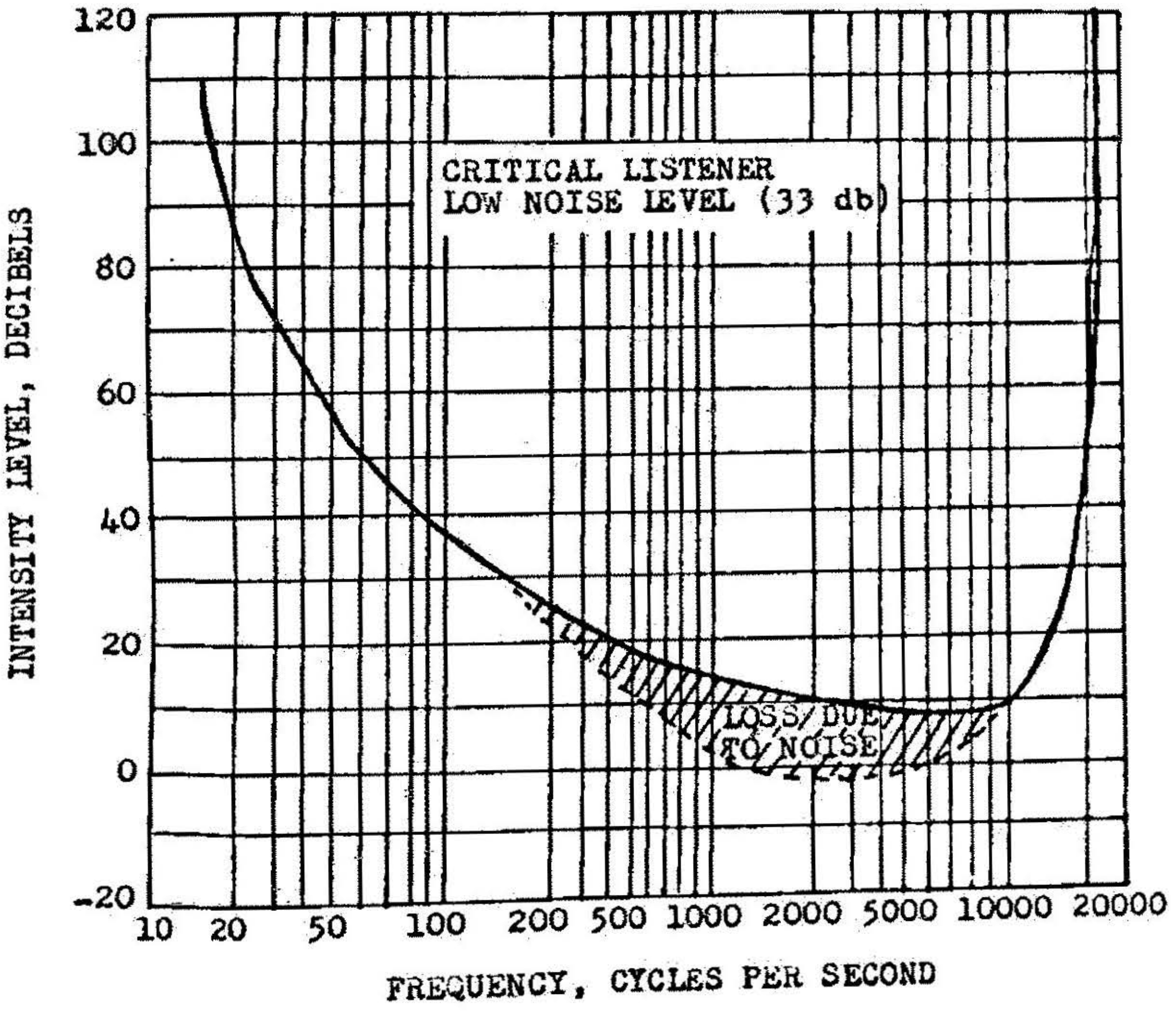

FIGURE 3*

EFFECTIVE HEARING CONTOUR FOR CRITICAL LISTENER

IN LOW NOISE LEVEL

(JENSEN, "FIGURE 5")

* Jensen, p. 5.

> listener to hear in the presence of representative noise conditions. For our purposes, we are interested in the average case which represents a very large segment of the population, and this results from pairing average hearing acuity with the masking contour corresponding to average residential noise. Although it probably involves much less than 1% of the population, we should also examine the case in which the combination of hearing acuity and masking contour yield the widest possible perceived frequency range. We may take the acuity of our previously defined critical listener and pair it with the masking for 33 db noise for this case. The contours for these two cases are shown in Fig. 4 and 5.[4]

All of these measurements of hearing acuity were taken at near-pain intensities in order to ascertain the widest limits of aural perception. At normal listening levels, the human ear is much less sensitive and of course, extraneous noise would influence the test except under laboratory conditions. Figure 4 shows the startling difference between the aural abilities of the critical and average listener.

> It is immediately apparent that . . . the listener is able to perceive only a restricted range of frequencies. For the average listener (at average reproduction level) the range of 175 to 5,800 cycles per second represents only about 62% of the total number of octaves assumed transmitted in the whole range of 40 to 15,000 cycles per second. The critical listener has an evident advantage for under the assumed conditions, he is able to perceive a range of 120 to 12,000 cycles which represents about 81% of the total number of octaves transmitted. The frequency range is greater for higher levels and lesser for lower levels as indicated in Fig. 8. The broken bars indicate the obliteration of a part

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[4] Jensen, pp. 4-5.

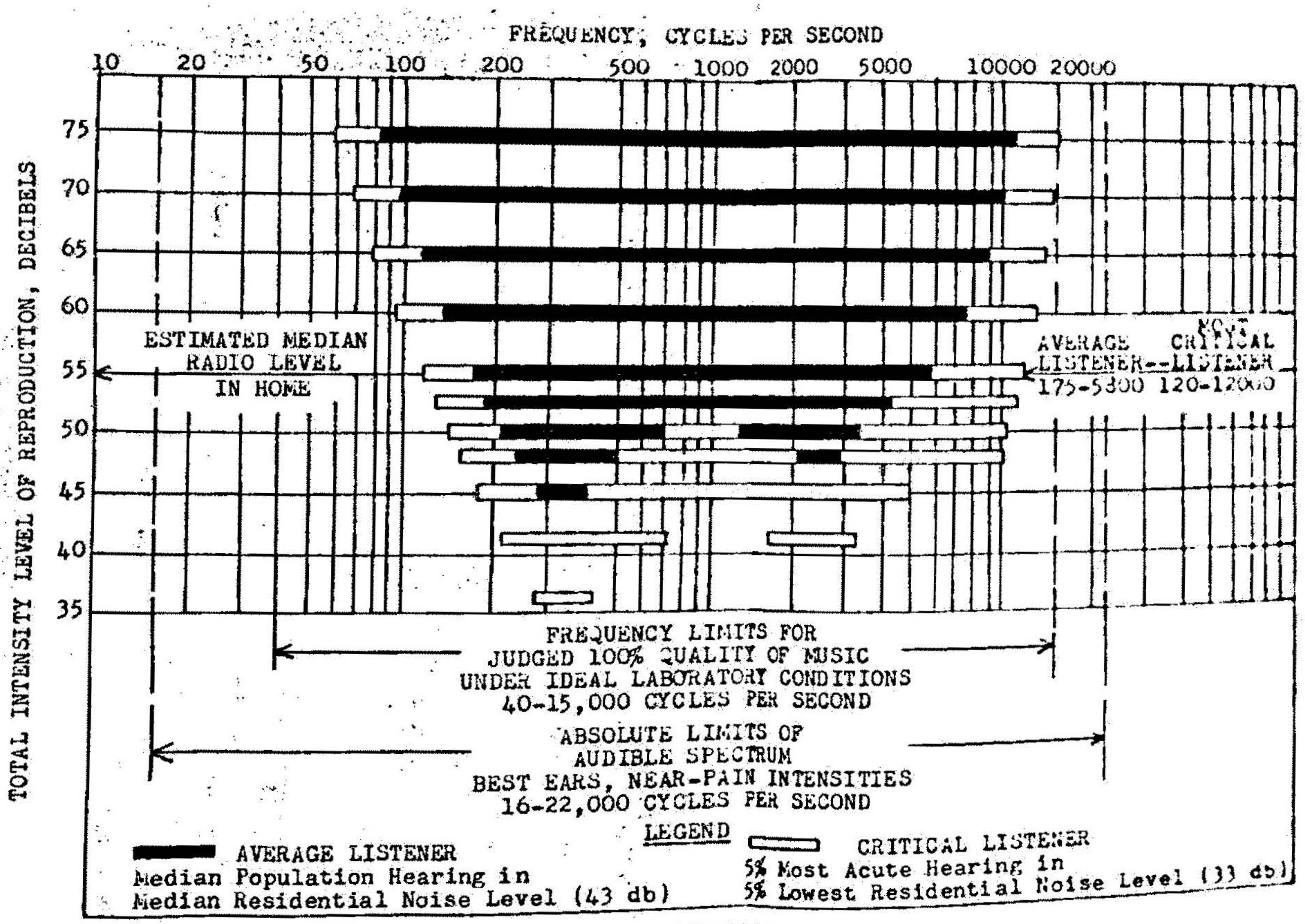

FIGURE 4*

STATISTICAL PERCEIVED FREQUENCY RANGES AS LIMITED BY HEARING AND NOISE. BASED ON MOST PROBABLE LEVELS IN CRITICAL FREQUENCY BANDS OF SYMPHONY ORCHESTRA MUSIC. (JENSEN, "FIGURE 8")

* Jensen, p. 7.

of the mid-range due to masking.[5]

From this study by the Jensen Manufacturing Company and other sources, we already can see some of the limitations of reproduced music as it reaches the ear of the listener. This study by the Jensen engineers assumed that the quality of the music being heard was a perfect reproduction of the original with no distortions, the highest fidelity equipment, and carefully controlled conditions. Perhaps we should now survey some of the problems attendant to the music itself.

To describe music in physical terms is exceedingly difficult. In symphony orchestra music, we know from tradition the kind and probable number of instruments which will be played, but the compositions and conducting technique introduce almost unlimited variables in frequency and intensity. We know that the great majority of existing radios and phonographs have a non-uniform response characteristic and reproduce a limited frequency range. This means that certain tones reproduced will be louder or softer than they were in the original sound and that all of the overtones that make up the quality of each tone will not be present in the reproduced sound. These facts make it quite possible, under certain conditions, for the listener to believe he is

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[5] Jensen, p. 6.

hearing a flute when he is actually hearing an oboe, because the overtones which give each instrument its characteristic quality are either not there or are so distorted as to confuse the listener. The last graph points out that the frequency response limits for judged one hundred per cent quality of music extends from 40 cycles per second to 15,000. This range of frequency would provide our ears with all of the sound we could normally expect them to perceive, assuming we have exceptional aural sensitivity at all frequencies. (Less than one per cent of the population possess hearing this efficient.)

Even though such a small percentage of the population can perceive the limits judged for one hundred per cent quality, the possible variables in reproducing and listening conditions make it advisable to keep this standard as one of our goals. The only justifiable reason for compromising on adequate frequency response would be for reasons of economy. The cost of electronic equipment rises sharply with its increased efficiency.

Commercially made equipment of all types to record and reproduce music are readily available in the form of amplifiers, speakers, radio tuners, tape recorders, etc., with high standards of efficiency and quality. The growing demand for this type of equipment by the two minority groups mentioned previously has encouraged manufacturers to design and market units to meet exacting standards and at a gradually reduced cost to the consumer. Periodicals and books are being published at an ever-increasing rate dealing with "High-Fidelity" in all its phases. Where radio magazines ten years ago would devote one page to "sound news" or "audio ideas," now whole magazines are published on the subject. The reader's problem is certainly not one of finding adequate equipment or information. The main problems lie in understanding how to use the equipment and, even more important, how to use it to reproduce music in a manner that is musically and artistically satisfying to the listener.

Unfortunately, much of the reproduced music we hear today is produced more by mechanical than by artistic standards. Radio stations are notorious for the manner in which they "monitor" their broadcasts. If the music gets loud enough to push the ever-present decibel meter up toward the "over modulation" mark, the engineer immediately turns down the "gain" or volume control. When the music softens, he turns it up again to maintain a good average modulation (i.e., degree of loudness) and avoid complaints from the listeners. This common practice results in the well-known "canned music" effect where the vitality of dynamic changes and effects have been mostly removed from the music and it tends to become more or less a background to other activities. The passive acceptance of this kind of reproduced music is the "average listener" at his worst. It is discouraging to realize that most commercial radio is aimed at pleasing only the average listener.

In fairness to the radio and recording engineer, it should be pointed out that it is at present impossible to reproduce the full dynamic range of an orchestra or chorus without reducing the loudest passages. This is to avoid distortion through exceeding the capacity of the equipment. It even may be necessary to amplify artifically the softest passages in order to over-ride the level of the ambient noise background. In addition to these two points, if the full dynamic range of an orchestra playing in an auditorium were reproduced in the average living room at the same dynamic level, the resulting sound would be over-powering and unpleasant. The solution to these problems lies in the knowledge and musicianship of the person or persons operating the sound reproducing equipment. The music must be heard at dynamic levels in proportion to the original sound and in proportion to the environment of the listener. Suggestions for achieving these proportions are discussed in detail in Chapters III and IV. There are many electronic instruments which can be used as aids in solving such problems, but in the final analysis, the success or failure of the methods used will be decided by the aural, musical, and psychological reactions of the critical listener.

The psychological reactions of the critical listener toward reproduced music must not be overlooked. Average listeners and even the most critical listeners are so accustomed to the inadequate fidelity of commercial radios, with their lack of sound presence and dynamic contrasts in the reproduced music, that they sub-consciously react against "high-fidelity" because its crisp aliveness and "real presence" makes it seem obtrusive by comparison. The writer once played a recording of a solo cello in the presence of several professional musicians, including the performer. The recording was clean and clear, and the cello sounded almost as though it actually were being played in the room. However, one of the listeners soon complained that the instrument sounded too "noisy and scratchy" to be lifelike. The writer re-assured the listener that the cello itself would seem even more noisy and scratchy because of the additional overtones present in the sound of the instrument itself. The recording equipment was incapable of duplicating these overtones. This listener would not believe this to be true until the original performer played his instrument in a side-by-side comparison with the recording. The indicated difference was at once apparent to all those present.

Another psychological problem is created when the listener cannot see the performer while listening to the music. The famous organist Virgil Fox made the following statement in a conversation after one of his concerts. "I believe the average audience is eighty per cent more responsive to my tour concerts when they can see me actually performing at the console of the organ." Even if Mr. Fox is only partially accurate in his statement, it becomes obvious that the listener is at a psychological disadvantage if he is listening to music over radio, phonograph, or tape recorders where he has lost the visual contact with the performer.

A second facet of the same problem has to do with the listener "participating" with the artist as he plays. It is quite possible for a listener to become so involved with the soloist and the music that he literally "breathes with the performer" and can be moved to tears or exultation by the performance. It is more difficult for the listener to "participate" in this manner when the music being heard is known to be simply a "re-creation" of a performance already "created" and the artists have since moved on to other tasks. The excitement of the concert hall and the live "audio-visual" performance has been removed. All that is left is a re-created version of the audible portion of the performance and the imagination of the listener. Chapter IV suggests some ways in which the imagination of the listener might be used to better prepare the environment surrounding a potential listening experience and help bridge these two important problems in the realm of the visual and "listener participation." Television seems to offer one of the better solutions to the visual participation of the listener because of the realization that in a "live show" he is actually watching the performer in action as he would in a concert hall. Unfortunately, most television stations concentrate on the quality of the picture being broadcast and the quality of the sound is rather inconsistent. Most television receivers on the market today have very inadequate audio systems, mainly because of their use of small, inexpensive speakers and the improper baffling of the speaker within the cabinet design. At the present time, the medium of sound reproduction which seems to meet most of the problems confronting the critical listener seeking a really satisfying listening experience is sound motion pictures. Well-recorded sound on film possesses excellent high-fidelity characteristics with almost no background noise, providing the projector is well-designed and it is placed in a separate room to eliminate its inherent mechanical noise during operation. Picture quality is excellent with full, natural color possible, and projected images can be large and life-like even in the home. The possible developments, present and future, in this medium are discussed in Chapter V.

Earlier in this chapter, it was stated that the writer would have difficulty communicating ideas to the reader without a mutually understandable vocabulary of technical terms. It is possible that the reader has encountered already a number of terms whose meanings and implications are not entirely understood. The remainder of this chapter has as its purpose the explanation of terms that need to be understood by the reader attempting to make use of this thesis. These terms fall into three major areas, namely, electronics, acoustics, and music. We will consider first the areas of acoustics and music which have to do with the nature of aural musical sound before electronic devices record or reproduce it.

A musical tone is said to have a certain (1) pitch, (2) intensity, (3) duration, (4) quality, and (5) timbre. Since all sound as we conceive it is produced by vibration, we simply measure the number of vibrations per second of a musical tone to determine its pitch. The musical note "A" above middle "C" on a piano keyboard is generally accepted as standard pitch at 440 vibrations per second. The intensity or amplitude of the vibration determines its loudness. The length of time the object continues to vibrate determines the duration of the tone. To describe the quality of a tone, we must first describe overtones and their aural effect. The true pitch of a tone is somewhat complicated by the fact that very few instruments produce a pure sound. The "A" string referred to on a piano, once struck, produces not only the pitch "A" at 440 vibrations per second; but the moment the string begins to vibrate its entire length, it also breaks up into secondary vibrations of smaller length, much like small waves forming on top of larger waves at the ocean beach. For example, each half of the string sets up a vibration just twice as fast as the fundamental vibration of the entire length of the string. These secondary vibrations being just twice as fast as the fundamental produce the pitch of "A" one octave higher, or 880 vibrations per second. The string continues to divide into smaller and smaller vibration ratios, producing various and higher pitches, but with less and less intensity. These secondary pitches are called overtones or harmonics because they are heard above the fundamental pitch. The overtones produced by a vibrating string fall into a definite pattern which is often called the "chord of nature" because of the harmonious manner in which they blend with the fundamental tone.

The quality of a tone may be said to consist of the sound that results from the blending of the fundamental and overtones of a given note on an instrument. There is more to it than this, however. The size, construction, and method of producing the initial tonal vibration in a given instrument have a marked effect on the intensity and order of these overtones. For example, a flute and an oboe can both play the same pitch as the piano, namely, "A" at 440 vibrations per second. The flute tone is produced by blowing across a hole in the mouthpipe while the oboe tone is produced by a double cane reed. These differences, coupled with the varied shape and construction of the two instruments, gives to each its particular pattern and intensity of overtones and thus that individual quality of sound associated aurally with each instrument is heard and identified. The timbre of a musical tone may be defined roughly as a varying combination of the tone's intensity, quality, and resonance.

Resonance is usually an important factor within the musical instrument itself. The volume of a piano tone is determined mostly by the resonating sound board in the instrument. The string sets the sound board into motion which in turn sets the air surrounding it in motion. The larger surface area of the sound board serves to amplify greatly the vibrations produced by the metal string. The tone from the sound board moves out and away from the piano in all directions and is reflected from the walls, floor, and ceiling back toward the listener again and again. These reflections add additional resonance to the tone which actually may continue to be heard after the piano itself has ceased to sound. Other objects in the room may begin to vibrate in sympathy with the piano tone and in this way add their own sounds to the total aural impression. A choir on tour and its director must face constantly changing physical environments as different auditoriums are encountered. The timbre of the sound produced by the ensemble is often changed radically by the reflection and absorption patterns that make up the "acoustics" of that particular room. These changes often have a noticeable psychological effect on the performers. Chapter IV discusses some of these effects and their importance to the critical listener as well as the performer.

At this point, it would be well to clarify the terminology used by the musician in contrast to that used by the acoustical engineer. The musician relies mostly on psychological terms in describing musical sound while the acoustical engineer will rely mainly on physical terms. Clear distinctions always must be drawn between physical terms and their psychological counterparts. Tonal attributes may then be summarized as follows: (1) All vibration has three physical attributes (a) frequency, (b) amplitude, and (c) vibration form. (2) These give rise in the mind of the musician to the three psychological attributes, or tonal attributes (a) pitch, (b) loudness, and (c) tone quality. Physical phenomena can be accurately measured, at least in theory. Psychological phenomena, being but sensations of the brain, are not subject to physical laws, can be only roughly measured (if at all) and often vary in very complex ways as the physical stimuli are varied even in a simple manner. Table I lists some psychological terms and their physical counterparts.

It would, of course, be possible to go on into detailed descriptions of other acoustical and psycho-acoustical phenomena, such as the harmonic theory of tone quality versus the format theory, difference tones, vibrato, decibel measurements, etc. However, in the interest of brevity, it seems best to limit the descriptive material included here to that which is most essential to an understanding of the immediate problems at hand. The reader desiring more detailed and comprehensive information in the areas of acoustics and music can find it readily through the use of the bibliography included with this thesis.

In the field of electronics, we find descriptive terms used freely from both acoustics and music, as the electronic engineer must use knowledge from these areas to better understand his specialized problems. The function of the electrical equipment necessary to record and reproduce music is that of changing the sound vibrations into audio frequencies and/or radio frequencies, of amplifying them electrically, and, through a loudspeaker, of changing them back into audible sound with as much fidelity to the original sound as desired by the listener.

TABLE I

PSYCHOLOGICAL TERMS AND THEIR PHYSICAL COUNTERPARTS

PSYCHOLOGICAL TERMS (Sensations)	PHYSICAL TERMS
1. Pitch depends on	fundamental or predominating frequency.
High tones caused by	high frequencies. Usually produced by small bodies.
Low tones caused by	low frequencies. Usually produced by large bodies.
2. Loudness depends on	amplitude, or physical "intensity."
Loud tones caused by	vibrations of considerable force and extent.
Soft tones caused by	vibrations of small force and extent.
3. Tone quality depends on	vibration form (the number of overtones and the way the energy is distributed among them).
Musical tones caused by	relatively regular and periodic vibration.
Pure tones caused by	simple vibration form.
Richer tones caused by	more complex vibration forms.
Brilliant tones caused by	very complex vibration forms.
Noise caused by	relatively irregular and non-periodic vibration, often with inharmonic overtones present.

* Wilmer T. Bartholomew, *Acoustics of Music* (New York: Prentice-Hall, Inc., 1942), p. 20.

Fidelity and distortion. Mr. F. Langford Smith of Australia has written a concise definition of fidelity and described various types of distortion in the following statement:

> True fidelity is perfect reproduction of the original. Distortion is due to the addition of features not in the original or the absence of features present in the original. Distortion may be classified under a number of headings:--
>
> 1. Harmonic distortion (the production of harmonics [or overtones] not present in the original).
>
> 2. Frequency distortion (unequal amplification of all frequencies [or pitches]) . . . .
>
> 4. Scale distortion . . . [usually due to the operation of the loudspeaker at a volume level other than that of the original sound. May be corrected with adequate tone controls.]
>
> 5. Distortion of transients. [a serious type of distortion which is noticeable through the hangover effect following a percussion noise. A very wide frequency range is essential for realistic reproduction of transients, also called overtones or harmonics.]
>
> . . . .
>
> If certain harmonics are unduly stressed or suppressed in the reproduction, the character of the sound will be changed. For example, it is possible for a displeasing human voice to be reproduced, after passing through a suitable filter, so as to be more pleasing, or vice versa. True fidelity, however, is fidelity to the original.[6]

In addition to the elimination of the various types of distortion just mentioned, the electronic equipment must

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[6] F. Langford Smith, editor, The Radiotron Designer's Handbook (Sydney, Australia: The Wireless Press, 1940), Distributed in U. S. A. by R C A Manufacturing Company, Inc., New Jersey, p. 31.

be capable of a sufficiently high maximum undistorted power output and a sufficiently low level of residual noise (hum, etc.) to give the required dynamic range.

Tone compensation and tone control. It is now necessary to survey tone control requirements as described by Mr. F. Langford Smith.

> An "ideal" audio frequency amplifier is one having a response which is linear and level over the whole audio frequency range. It is sometimes desirable to control the frequency response so as to compensate for certain non-linear components such as pickups or loudspeakers, or to enable the listener to adjust the tone to suit his taste. Such methods of control are known by many names, such as Tone Compensation, Tone Control, and Bass Boosting. [And Treble Boosting or Attenuation.]
>
> . . . .
>
> Comparatively simple combinations of resistance, capacitance and inductance may be used, in conjunction with amplifying valves, [tubes] to produce a fairly wide range of frequency characteristics.[7]

Adequate tone controls are essential to any music reproducing equipment because of the constant variations encountered in the types of music being reproduced and the marked difference in the acoustics surrounding each performing group--some in auditoriums, some in studios, and some perhaps even out-of-doors. Mr. Smith concludes his description of tone control requirements as follows:

> If the whole of the equipment from microphone to loudspeaker inclusive has a uniform frequency response, the reproduction will only sound natural if the acoustic

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[7] Ibid., p. 58.

> level of the reproduction is the same as that of the original sound. At lower levels, due to the characteristics of the human ear, there will be a pronounced drop in the apparent bass, and a less pronounced drop in the apparent treble. If the closest possible approach to the impression of the original sound is required at a lower acoustic level, it is necessary to provide the requisite [adequate] degrees of bass and treble boosting.[8]

Volume expansion and compression. The reader may recall the statement earlier in this chapter (page 12) concerning the impossibility of recording the full dynamic range of an orchestra without exceeding the capacity of the recorder. The recording apparatus must be monitored in some fashion. It is possible with volume compressor and expander circuits to meet this problem electronically with considerable success.

> It is possible in an amplifier to vary the gain in such a manner that the greater the input the greater the gain [volume]. Such procedure is called volume expansion and the reverse action is called volume compression. It is not possible here to comment on the desirability of such systems since many factors have to be considered. Various types of expansion or contraction characteristics can be produced and a wide variation of the time constant can be obtained. In general a time constant of about one-fifth of a second is usually considered reasonably satisfactory, but with elaborate amplifiers it is possible to obtain a more rapid pickup and a slower decline.
>
> . . . .
>
> Volume compressors are somewhat similar to volume expanders except that the action is inverted.
>
> All existing types of volume expanders and compressors necessarily need to be a compromise. If it is

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[8] Ibid., p. 70.

desired to make the received sound identical with that in the studio, it is necessary to have identical contraction and expansion characteristics, and zero time delay. This can only be obtained or approached closely by a system in which a monitor signal is transmitted in order to control the gain in the reproducing amplifier to correspond with the compression in the studio.[9]

Phonograph records. Phonograph records are subject to a wide variation of frequency response and over-all quality from manufacturer to manufacturer. Some of the problems are mentioned in the following quotation:

> Standard "lateral-cut" gramophone [phonograph] records are recorded with a characteristic in which the amplitude of the needle movement is proportional to the input voltage for frequencies above 250 c/s. Due to the fact that for constant acoustic power the amplitude increases as the frequency is decreased, a point is reached beyond which there is danger of one groove cutting into the next. Consequently below 250 c/s. the recording is made to follow a "constant Amplitude Characteristic," which means that the same amplitude holds for a given applied acoustic power at all frequencies. This is equivalent to a drop in output of 6 db. per octave (1 octave = frequency ratio 2 : 1), or to a voltage ratio of 2 : 1 per octave. The diagram (Fig. 1) shows the theoretical output given by an ideal uncompensated pickup, and also the compensation required (upper curve with broken line) for overall level response.[10]

Phonograph pickups. The phonograph pickup is a good example of the manner in which various parts of a music reproducer can effect the quality of the reproduced sound. One unit may compare in price and quality very closely with another and yet, due to the problems listed below, produce

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9 Ibid., p. 74.

10 Ibid., p. 75.

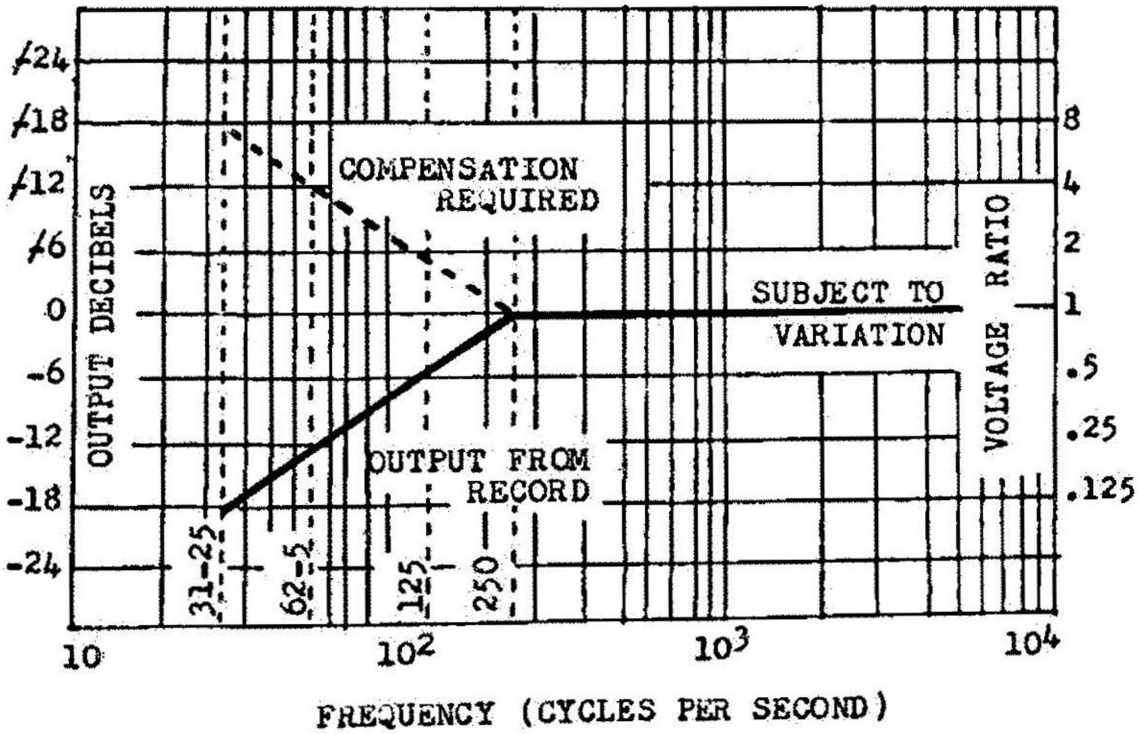

FIGURE 5*

THEORETICAL OUTPUT GIVEN BY AN IDEAL UNCOMPENSATED PICKUP, AND COMPENSATION REQUIRED FOR OVERALL LEVEL RESPONSE

(SMITH, "FIGURE 1")

* Smith, p. 75.

a noticeably different quality of sound. Mr. F. Langford Smith describes pickup faults in the following quotation:

> There are many types of pickups in use. . . . All types, including crystal types, suffer to a greater or less extent from the following:--
> (1) Mechanical resonances of the pickup either in part or whole.
> (2) Mechanical resonance of the arm and pickup as a whole.
> (3) Harmonic distortion of the output due to non-linear relationship between the movement of the needle and the output voltage.
> (4) Non-uniform output voltage at all audible frequencies.
> (5) Wear of the record due to weight, poor tracking, mechanical damping, etc.
>
> Mechanical resonances at the higher audio frequencies are objectionable, and may only be reduced by a modification in the internal structure of the pickup. Nearly all pickups employ some form of damping (usually mechanical) to reduce these resonances to some extent, but complete elimination is not practicable owing to wear caused on the record.
>
> Resonance of the arm as a whole is unavoidable, and is frequently used in popular types to improve the bass response, thereby giving some compensation for the recording characteristic.[11]

Microphones. Microphones, like phonograph pickups, are subject to definite mechanical limitations, and various types of microphones often will change the quality of the music they reproduce without necessarily being inefficient. Mr. Smith gives the following description of various types of microphones:

> Microphones may be divided into two groups:

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11 Smith, pp. 75-6.

(1) Low impedance
(2) High impedance

In the first group are various types of carbon or granule, velocity (or ribbon), and dynamic microphones. In the second group are condenser and crystal microphones. The latter may be subdivided into:

(a) Diaphragm types, giving greater output but having poorer fidelity, and

(b) Sound cell types in which the air pressure operates directly on the crystal face.[12]

Mr. Leo Beranek, an acoustical engineer describes pressure microphones as follows:

> Pressure microphones are the most widely used of the three basic types discussed in the preceding part. They are applicable to acoustic measuring systems and to the pickup of music and speech in broadcast studios, in public-address installations, and in hearing aids. Many engineers and artists believe that music reproduced from the output of a well-designed pressure microphone is superior to that from the more directional types of microphone because the quality of the reverberation in the auditorium or studio is fully preserved, because undesirable wave-form distortion is minimized, and because the quality of the reproduced sound is not as strongly dependent as for other types upon how close the talker or the musical instrument is to the microphone.[13]

Tape and wire recording heads. A steel wire or some type of tape coated with a magnetic material is pulled through a dual magnetic field. The first field is vibrating at a constant frequency above the range of audibility. This tone pulls the molecules on the tape or wire in a uniform direction which eliminates most residual noise and any previous recording. The second magnetic field varies in

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[12] Ibid., p. 77.

[13] Leo L. Beranek, Acoustics (New York: McGraw-Hill Book Company, Inc., 1954), pp. 150-1.

frequency with the sound being recorded which changes the molecular structure of the tape or wire accordingly. Pulling the recorded tape or wire through a third magnetic field induces changes in the field in direct ratio to the frequency of the molecular changes in the wire or tape, thus reproducing the original sound. The advantages of this type of recording are a low background noise, a wide dynamic range, and excellent fidelity in high quality units. A disadvantage is the necessity of pulling the wire or tape across the heads at an absolutely constant speed to avoid "wow" and flutter. The fact that the fidelity increases with the speed of the pull thereby limits the length of recordings to convenient spool sizes.

Loudspeakers and their enclosures. The subject of loudspeakers and enclosures is now introduced through the writing of Mr. Beranek in his new book called Acoustics.

> A loudspeaker is an electromagnetic transducer for converting electrical signals into sounds. There are two principal types of loudspeakers: those in which the vibrating surface (called the diaphragm) radiates sound directly into the air, and those in which a horn is interposed between the diaphragm and the air. The direct-radiator type is used in most home radio receiving sets, in phonographs, and in small public-address systems. The horn type is used in high-fidelity reproducing systems, in large sound systems in theaters and auditoriums, and in music and outdoor-announcing systems.
>
> The principal advantages of the direct-radiator type are (1) small size, (2) low cost, and (3) a satisfactory response over a comparatively wide frequency range. The principal disadvantages are (1) low efficiency, (2) narrow directivity pattern at high frequencies, and (3) frequently, irregular response curve at high frequencies.

For use in home radio receiving sets where little acoustic power is necessary and where the listeners are generally not very critical, the advantages far outweigh the disadvantages. In theater and outdoor sound systems where large amounts of acoustic power are necessary and where space is not important, the more efficient horn-type loudspeaker is generally used. . . . In this text, [thesis] however, we shall limit ourselves to moving-coil loudspeakers, the type commonly used in radios and home music systems.[14]

Horn-type loudspeakers eventually may replace all direct-radiator-type reproducers as they can be constructed to fit into the actual design of a house, using, for example, the walls of the room itself as an extension of the horn.

By definition, a baffle is any means for acoustically isolating the front side of the diaphragm from the rear side.[15]

Many new homes incorporate direct-radiator speakers into the construction of a wall to isolate the vibrations from the rear of the diaphragm completely and direct them into some other part of the house. Mr. David Fidelman describes the basic principles of baffle design in the following quotation:

The reasons for the difficulty in obtaining proper baffling for loudspeakers may not be immediately apparent until it is realized that the primary purpose of the baffle is to prevent sound from the back of the speaker cone (which is 180° out-of-phase with that from the front) from cancelling the sound transmitted from the front of

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14 *Ibid.*, p. 183.

15 *Ibid.*, p. 184.

the cone. The smaller the difference in air path compared to the wave-length of the sound, the more complete is the cancellation. Therefore if the loudspeaker is not mounted in a baffle, or is mounted in a very small one, there will be cancellation up to a relatively high frequency and the reproduced sound will be deficient in low frequencies. If the speaker is mounted on a flat board, which is the simplest type of baffle, this board should measure at least 8 feet on each side (with the speaker mounted away from the center) if adequate response is to be obtained below 100 cps.

Because of the large size required if flat boards are used for loudspeaker mountings, a number of different types of baffles have been developed which do not require as much space. Some types perform the additional function of improving the low-frequency response by increasing the coupling between the loudspeaker cone and the air into which the sound is radiated. At low frequencies the area of the loudspeaker becomes insufficient for proper coupling to the air--this is one reason why small loudspeakers are not capable of the same low-frequency response as larger loudspeakers. The enclosures which increase the low-frequency response do so by increasing the area of radiation into the air at low frequencies.

The most common type of housing for loudspeakers is the unsatisfactory conventional open-back cabinet found in almost all commercial radio and radio-phonograph combinations sold at the present time. When the sound path from the back of the cone is sufficiently long (as in the case of the large console cabinets) the low frequencies are reproduced; while in the midget radio cabinets the sound path from the back to the front is very short and the low frequencies are not reproduced because of the out-of-phase cancellation. However, the most objectionable acoustical feature of such cabinets is that the back of the cabinet behind the loudspeaker acts as a resonant enclosure. It is an open-ended resonant tube (such as, for example, an organ pipe) which accentuates the loudspeaker response at the frequency of resonance due to the increased efficiency of the acoustical system. . . . The cabinet resonance causes the sharp peak in the response, generally in the range between 100 and 200 cps, which very unfavorably affects the intelligibility and naturalness of the reproduced sound and is especially noticeable in the reproduction of music and male speech. This is the "boomy" quality so characteristic of almost all commercial radio receivers. This

> open-back construction of the loudspeaker enclosure is used in mass-produced receivers because of its low cost and simplicity of construction, but it should be avoided in any system being set up for high quality reproduction.[16]

Fortunately, radio manufacturers are becoming more careful in cabinet construction and are attempting to design cabinets which properly baffle the loudspeaker in addition to housing the mechanical equipment. Mr. Fidelman goes on to describe the infinite-type baffle.

> The simplest type of loudspeaker cabinet is one with a completely enclosed back. By making the cabinet as rigid as possible and padding the inside with absorbent material, the sound from the back of the loudspeaker cone is completely prevented from reaching the front. Such a cabinet is sometimes known as an "infinite baffle" cabinet, since its effect is similar to mounting the loudspeaker on an infinitely large flat board. However, the volume inside the box must be sufficiently large, or else the low-frequency response will be reduced.[17]

The most popular type of baffle and the one most easily constructed and adjusted to fit the room environment is the bass-reflex cabinet. It is described by Mr. Fidelman as follows:

> It is very simple to construct and, when properly designed, gives excellent acoustic results. Many manufacturers provide such cabinets for use with their loudspeakers, and they have been used commercially for loudspeakers ranging in size from 8-inch to the large 18-inch low-frequency units of dual systems used in theaters and auditoriums.

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16 David Fidelman, "Loudspeaker Enclosures," Radio & Television News, 47:50, June, 1952.

17 Loc. cit.

The basic principle of operation of the bass-reflex cabinet is the use of acoustical networks to increase the low-frequency response of the loud-speaker. The construction. . . consist[s] of a closed cabinet with an opening in the front close to the loudspeaker. The volume of the cabinet has the properties of an acoustical capacity, while the opening in the front has the properties of an inductance in series with the acoustic resistance of the air. . . . The low-frequency response is increased by the coupling of the two tuned circuits, because the currents in the two resistors are in phase, and the sound from both the front and the back of the cone is therefore useful.

However, good results are obtained from the bass-reflex cabinet only when it is properly designed to match the size and resonant frequency of the loudspeaker with which it is to be used. Improperly designed cabinets will produce undesirably boomy and resonant bass, therefore the experimenter who constructs his own bass-reflex cabinet should be careful to use proper dimensions in his construction.[18]

The acoustical labyrinth cabinet is quite difficult to construct but provides excellent results.

In this type of enclosure, the acoustic tuned circuit of the bass-reflex cabinet is replaced by a resonant line. An absorbent-walled tube is coupled to the back of the loudspeaker at one end, and is open to the air at the other end. At the frequency for which it is one-quarter wavelength long, this tube sees a low impedance at the open end, and therefore presents a high impedance to the back of the loudspeaker cone. Thus, by choosing the length of the tube so that it is a quarter-wave-length at the resonant frequency of the loudspeaker suspension, the resonance of the speaker is damped in the same manner as with the bass-reflex cabinet. At double the resonant frequency, the tube is a half-wave-length long and the phase is reversed, therefore the sound through the tube is in-phase with that from the front of the loudspeaker and the response is increased. The tube lining absorbs almost all of the sound above 150 cps therefore the higher resonances have no effect.

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18 Ibid., p. 55.

In the labyrinth cabinet, this resonant tube is folded so that the total outside dimensions are practical for use in the home or studio.[19]

The following cabinet design also can be incorporated into the actual construction of a home or auditorium and therefore is more expensive but provides excellent results.

A type of loudspeaker cabinet which is becoming widely used because of its good low-frequency response is the folded-horn cabinet. In this type of cabinet the sound is radiated from the front of the speaker cone at high frequencies and through a horn coupled to the back of the cone at low frequencies. For home use, it is generally designed to be placed in a corner of the room so that the walls and floor form part of the horn.

The horn is used in loudspeaker applications because it is the acoustical equivalent of the electrical transformer. Since at low frequencies the air represents too low an impedance for proper coupling to the loudspeaker cone, the horn can be used to transform this low impedance to a higher impedance which permits more efficient energy transfer. Generally a volume of air is maintained between the loudspeaker and the entrance to the horn, to act as an acoustic capacity which bypasses the horn at higher frequencies so that all the high-frequency sound radiation is from the front of the speaker. With the use of properly designed corner folded-horn cabinets, frequencies as low as 20 to 30 cps can be reproduced using standard commercial loudspeakers in cabinets of practical sizes.

The details of construction of a typical corner folded-horn cabinet are shown on the "data-print." Practical dimensions are given for the construction of such cabinets for use with commercial 12-inch and 15-inch loudspeakers. No dimensions are given for use with smaller speakers, since the major usefulness of the horn is with speakers that are capable of producing the very low frequencies at which the horn coupling to the air

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[19] _Ibid._, pp. 98-99.

is most useful.[20]

We are indebted to Mr. Fidelman for his competent descriptions of various baffle enclosures.

<u>The horn-type loudspeaker for extended range systems.</u> The direct-radiator loudspeaker is often called the "woofer" because its main responsibility in a two-way speaker system is to produce the low frequencies with the aid of proper baffling. The high frequency horn-type speaker is often called the "tweeter" because of its high pitched shrillness when heard by itself. Higher frequencies can be produced, and produced more efficiently, by the horn-type tweeter than they can by the cone-type direct-radiator speaker. When one speaker, or more, of each type is used, extremely wide range reproduction is possible. Frequency dividing networks are necessary to provide each speaker unit with the band of frequencies it is designed to reproduce. Mr. Beranek describes horn-type speakers in considerable detail, part of which is included here.

> The driving unit for a horn loudspeaker is essentially a small direct-radiator loudspeaker that couples to the throat of a flaring horn.[21]
>
> When well-designed, the large end of the horn, called the "mouth," has an area sufficiently large to radiate sound efficiently at the lowest frequency desired. The

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20 <u>Ibid</u>., pp. 98-99.

21 Beranek, <u>op</u>. <u>cit</u>., p. 260.

small end of the horn, called the "throat," has an area selected to match the acoustic impedance of the driving unit and to produce as little nonlinear distortion of the acoustic signal as possible.[22]

Sound "presence" and "high-fidelity." These two terms are often mis-used and mis-understood as they are applied descriptively to sound reproduction. Generally, reproduced sound is said to have "presence" if it is clear and crisp and gives the listener the aural impression of nearness. "Presence" is strictly a psychological term while "high-fidelity" should be used as a physical term. Unfortunately, however, the label "high-fidelity" is attached to almost anything that is in any way connected with sound reproduction. As a physical term, high-fidelity refers to the frequency response range necessary to reproduce sound in a manner completely faithful to the original. Note the following statement from engineers at the Jensen Radio Manufacturing Company:

> The term "High-Fidelity" deserves careful use. If the term is to retain any meaning, it does not seem to be proper to apply it to a band less narrow than No. 3 (75 to 8,000 cycles). The term "Medium-Fidelity" seems appropriate for Bands 4 and 5 (down to 110 to 5,300 cycles), while narrower bands, in view of the present state of the art are "Low-Fidelity" in their performance.[23]

Page 37 quotes a suggested frequency band chart by the

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22 Ibid., p. 259.

23 Jensen, op. cit., p. 12.

TABLE II*

A PREFERRED SERIES OF AUDIO FREQUENCY BANDS

FOR SOUND REPRODUCING SYSTEMS

(JENSEN, "TABLE 1")

Band Number	Classification	Cut-off Frequencies Low	High
1	High Fidelity	40	15,000
2	High Fidelity	65	11,000
3	High Fidelity	75	8,000
4	Medium Fidelity	90	6,400
5	Medium Fidelity	110	5,300
6	Medium Fidelity	130	4,400
7	Low Fidelity	160	3,600
8	Low Fidelity	200	3,000

A. Band 1 is the assumed complete spectrum of music. FCC requirements for FM transmission call for a range of 30 to 15,000, uniform within 2 db.

B. Band 2 affords as complete fidelity as Band 1 for a critical listener (5% most acute hearing) in very quiet homes (5% quietest, 33 db noise level) at usual reproduction levels.

C. Band 3 affords as complete fidelity as Band 1 for an average listener (median population hearing) in an average home (median annual noise level, 43 db) at usual reproduction levels.

D. Band 2 or 3 is approximate maximum range of high quality transcriptions.

E. Band 5 or 6 is approximate maximum useful range for nighttime and rural reproduction of AM broadcasting and commercial lateral phonograph records.

F. Bands 2 to 6 probably require console type radio receivers for reproduction of low end.

G. Aural balance will probably be acceptable if one cut-off is paired with that in an adjacent band. Thus: 65-90 to 8,000; 90-130 to 5,300, etc.

* Jensen, p. 11.

Jensen engineers concerning fidelity standards.

Electrical measuring devices. The human ear is not reliable as a measuring device in the physical sense. Therefore, mechanical and electrical instruments must be used for tests and measurements by the engineer and technician.

The oscilloscope and some of its applications. The oscilloscope is an electronic instrument which produces a visual "wave pattern" of any vibration form of audio or radio frequencies on the screen of a cathode ray tube. Through this instrument, it is possible to study overtone patterns and test radio and audio circuits for almost all kinds of distortion.

Decibel meters. A decibel meter measures the change of power flowing through a circuit. This measurement is helpful as a general visual indication of loudness. However, the decibel should not be confused with the unit of measurement for loudness called the "phon."

> There tends to be confusion between the Decibel which is not a loudness unit but a unit of change of power, and the Phon which is a true loudness unit. If the ear were equally responsive to all frequencies, the Phon and the Decibel would be identical. The Decibel bears no relation whatever to the frequency response of the ear but rather to the conditions existing in an electrical circuit. The Phon being a unit of loudness is directly affected by the characteristic of the ear and therefore there is no direct relationship between Decibels and Phons over the audio range.[24]

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24 Smith, op. cit., p. 88.

For descriptions of other electrical measuring instruments and their uses, see Chapter 32 of Mr. F. Langford Smith's The Radiotron Designer's Handbook (page 263). It must be remembered that in the final analysis, no instruments can replace the aural evaluation of the critical listener. The final objective of all electronic engineering in the area of music reproduction must be to make the music artistically and psychologically satisfying to the listener.

CHAPTER II

THE LITERATURE AVAILABLE AND THE LIMITATIONS OF PREVIOUS STUDIES

In Chapter I (page 11) some idea is given of the amount of literature available in the general subject area of this thesis. Periodicals and books are being published at an ever-increasing rate dealing with high-fidelity in all its phases. It does not seem necessary to face the reader with an exhaustive analysis of all the various studies dealing with acoustics, electronics and music in the area of music reproduction. However, to understand more adequately the types of literature available, we will survey specific references in each of the three general areas just mentioned.

The study of acoustics as it relates to musical sound reproduction is a very new field. Most references that are available were written as little as fifteen years ago and already are very much out-of-date in terms of new discoveries, particularly in the area of psycho-acoustical phenomena. Therefore, it seems best to use the limited space available in this thesis surveying the most up-to-date studies.

One of the latest texts is called Acoustics of Music by Wilmer T. Bartholomew. Mr. Bartholomew is a musician as well as an acoustical technician. This makes his book one of the best available as a fusion of the science of acoustics and the art of music. The following statement, taken from the preface of Mr. Bartholomew's book, reveals his thinking in these areas:

> The Taj Mahal, that dream of architectural loveliness, seems to our entranced vision far removed from such mundane things as the measurement of angles, the stresses and strains of building materials, and the chemistry of pigments. . . . Beethoven's Ninth Symphony, that last mighty dream of a stormy soul, . . . seems to our transported ears to have no connection with such prosaic things as the compressibility of air, the reflection of sound waves by walls, or the mathematics of Fourier analysis. But there is a connection, unrecognized and scorned though it be by many. We are usually unconscious of it during the emotional experience of listening to music. But whether or not we like to admit our dependence on such material things, there is no art capable of rearing high its magic towers of beauty without building them on the strong foundation of the established and immutable laws of "the-nature-of-things," or science. . . . An undiscriminating writer has said, "The soul of the piano transcends all investigation." Such a statement is patently incorrect, since the soul is not in the piano but in the pianist--and I hope it will there continue to transcend investigation. That which is essentially imponderable cannot be weighed or analyzed very far. But it is quite possible to study the actions of the muscles of the player, and draw certain conclusions that may greatly aid the pianist in the expression of whatever is in his soul.
>
> . . . .
>
> This book, therefore, is written from the musician's standpoint, in the hope of answering the questions that music students and performers might ask.[1]

From this brief excerpt of Mr. Bartholomew's preface, we gain the distinct impression that he is attempting what some might believe to be impossible; namely, that the indef-

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[1] Bartholomew, op. cit., pp. vii-viii.

inite and impossible to measure esthetics of music and the more definite and specific science of acoustics can be treated as a single subject area. Further study of his book reveals that in the main it relies heavily on the science of acoustics for its effectiveness and deals in a very secondary way with the related problems of music in the artistic sense.

In its early chapters, this book deals on a non-mathematical basis with the usual elementary acoustical problems of the nature of vibration, sound waves, vibratory sources of sound, with various sub-topics under each. The last three chapters are perhaps the most informative as they relate to this thesis. Chapter IV called "Harmony and Scales" deals with various problems of tuning (true or "just" intonation), equal temperament, etc. A study of this chapter by the reader would be very beneficial as an aid in understanding the problems singers and instrumentalists face in producing melodies and harmonies "in tune."

Chapter V deals with the mechanism of hearing. This chapter is an aid to the reader in comprehending limitations of the human aural mechanism. However, Chapter XIII of the book Acoustics by Leo L. Beranek covers this subject much more completely, and, therefore, is recommended above the Bartholomew chapter. Chapter VI of Acoustics of Music comes the closest to the problems presented in Chapter I of this thesis as it deals with "Electronic Recording, Reproducing, and Synthesizing of Sound." Unfortunately, the chapter is very brief and as such it serves more to list the problems rather than to discuss possible solutions to them.

As a conclusion to the discussion of the Bartholomew book, it can be said that this volume is one of the best studies available on the acoustics of music. Since the book is, in the main, non-mathematical in its approach and deals sympathetically with the musician's problems, the reader does not get lost in the complexities of formulas and logarithms. Acoustics by Beranek is a much more adequate text from the acoustical engineer's standpoint because it is based upon mathematics. In comparison to the Bartholomew text, the Beranek book is much less useful to the lay reader.

In the area of electronics, reference is made to the book by F. Langford Smith, entitled The Radiotron Designer's Handbook. This book is still adequate as a basic reference for information on standard definitions, radio frequency, and audio frequency circuits, even though considerably out-of-date in terms of the thesis subject.

It is interesting to note that when the writer began accumulating information for this thesis, in 1940, there were no commercially practical wire-recorders, tape-recorders, long-playing records, or television sets. This serves to illustrate that commercial developments have come so rapidly and technological studies have advanced so quickly that it is difficult, if not impossible, to find any single source of information that adequately covers the subject. Because of this fact, it becomes necessary to accumulate information on electronic developments from many varied sources. One of the most valuable sources is current periodicals. Research and new designs for experimental and commercial equipment are illustrated and discussed from week to week in many periodicals. The reader will note in the Bibliography the extended list of periodicals giving sources of information in the electronics area.

In addition to periodicals, there are many valuable technical monographs prepared by various industrial manufacturers in the electronics field. The Jensen Manufacturing Company of Chicago, Illinois, published a series of four Technical Monographs in 1945 dealing with the following subjects: Number One, "Loud Speaker Frequency-Response Measurements," Number Two, "Impedance Matching and Power Distribution," Number Three, "Frequency Range and Power Considerations in Music Reproduction," and Number Four, "The Effective Reproduction of Speech." The third monograph in this series has been most useful as a reference for this thesis and quotations will be found in Chapter I from this work.

Another example of a useful technical monograph is the Amplifier Manual by A. C. Shaney, Chief Engineer of the Amplifier Company of America. This monograph gives excellent illustrations of the manner in which radio engineers are designing amplifiers that attempt to solve problems of tone control, volume expansion and compression, scratch suppression, remote control, audio-spectrum control, etc.

In addition to periodicals and monographs listing new developments, there are important periodicals which can help the reader to evaluate more adequately the efficiency of the new equipment which is constantly made available to the public. Consumer Reports is a good example of such a periodical. This magazine is published by Consumers Union of the United States, Inc. The April 1954, issue, as an illustration, contains an extensive article covering the latest high-fidelity amplifiers on the market at that time. They are rated on the basis of cost, number of tubes, adequacy of frequency response, power, tone control, and available in-puts and out-puts. This type of information can be exceedingly valuable to the reader attempting to find first-rate components for a reproducing system.

An article in Life magazine, February 1955, quotes a group of unknown "experts" who have surveyed high-fidelity equipment currently on the market and make specific recommendations in terms of efficiency, brand name, and price. Supposedly poorer components have been weeded out, leaving only the problems of selecting the various features desired and noting the price limitations facing the buyer. Unfortunately, articles such as this grossly over-simplify the problems facing the consumer who might be a truly critical listener. The problems of construction and assembly of the various components and the techniques required to operate them are ignored completely. It is through these problems of assembly and operation of music reproducing equipment that one immediately becomes involved in a bewildering combination of the acoustical, electronic, and musical considerations, the solutions to which are the main concern of this thesis.

So far, we have considered some available literature in the fields of acoustics and electronics. The third aspect, namely, the field of music, has many excellent books dealing with its various phases. As a specific illustration, closely paralleling some of the musical problems faced in this thesis, there is the book by the eminent musician and conductor, Leopold Stokowski, entitled _Music for All of Us_. Mr. Stokowski has provided a book which can help the open-minded and discerning reader in understanding the subjective and objective ideas and emotions that course through the minds of musicians. Musicians are faced constantly with combinations of physical and psychological problems in their search for beauty in music. The book is a very personal and revealing expression of Mr. Stokowski's own ideals and beliefs concerning all phases of music. He has studied acoustics, the musical instruments of all nations and peoples, and he has observed, in most cases first-hand, the many modes of musical expression used throughout the world. Through these observations and study has come an understanding of the true depth of "The Soul of Music . . . The Heart of Music . . . The Physical Beauty of Music . . . The Mind and Music . . . Freedom of Response to Music."[2]

Through a better understanding of the soul, heart, and mind of music, the reader should be led to a greater freedom of response to music; this is obviously one of the major goals of this book. The present study also must face the problems certain individuals have in freely responding to music regardless of its medium of expression or methods of reproduction. Personal prejudices and previous conditioning experiences make it extremely difficult for some people to respond freely to reproduced music no matter how perfectly the reproduction is accomplished. This writer therefore recommends study of the Stokowski book as a requisite for all musicians and music students seeking new ideas and

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[2] Stokowski, op. cit., p. v.

concepts concerning their understanding and response to all kinds of musical expression. Chapter V of this thesis deals in some detail with the fascinating possibilities of new kinds of musical expression in the future through electric instruments of all kinds.

Mr. Stokowski writes of recorded and reproduced music in Chapters 28 through 33. The Chapter headings are (28) Recorded Music (29) Broadcast Music (30) Frequency Modulation (31) Music and Motion Pictures (32) Music and Television (33) Reproduction of Recorded and Broadcast Music. The first six chapters deal mainly with an outline of the latest developments in each field mentioned and suggest ways in which these developments can most benefit our daily lives. Chapter 32 lists some important generalizations for good volume, tone control, and sound dispersion techniques for better listening. This material is referred to specifically in Chapters IV and V of this thesis. The last part of Chapter 32 describes some esthetic and psychological possibilities of sound reproduction through acoustical and atmospherical control of the listener's environment, even to the inclusion of characteristic odors in the air. The purpose of this chapter is to open new and more comprehensive concepts of future developments in this field.

While there is much that is useful to both the layman and technical reader in this book, it is too general and incomplete in the more technical areas covered by this thesis. The book does not attempt to deal specifically with the problems of assembly and operation of reproducing equipment. However, in conjunction with the acoustical and electronical references previously discussed in this chapter, Mr. Stokowski's book is very valuable as an aid to the reader desiring to increase his musical awareness and sensitivity, as a counterbalance against becoming overly concerned with the more mechanical aspects of music reproduction.

The last example of a type of available literature to be surveyed here is the book by Carl Seashore, entitled <u>In Search of Beauty in Music</u>. A good introduction to the book is this statement in the Preface:

> More has been achieved in the laying of foundations for science in music in the present century than in all preceding centuries. The chief reason for this is that the applied science of music has had to await the development of such underlying sciences as acoustics, physiology, electrical engineering, anthropology, experimental education, and experimental psychology. In all these fields phenomenal progress has been made in the instrumentation and standardizing of techniques of measurement. Naturally to these should be added the development of a body of science-minded musical artists who welcome such scientific approaches. The progress has been facilitated and rushed at phenomenal speed by the practical aspects of radio, phono-photography, phonography, industrial acoustics, and the increasing demands for the psychology of music in these fields, as well as in education.
>
> . . . .
>
> This volume is designed as an introduction to the science of music for advanced students of music and psychology, music teachers, educators, professional

musicians, and general readers interested in the scientific approach to the understanding and appreciation of beauty in music.[3]

It is this book by Seashore that aided the writer most in finding a justifiable need for a more scientific approach to the problems of music reproduction. Note Mr. Seashore's statement contrasting the scientific approach with the laissez faire attitude:

> From the time of Aristoxenus and Pythagoras, there have been two basic attitudes toward music: one the impressionistic attitude of the musician who is not interested in explanations but merely in results which are judged by his unaided ear and speculative mind; the other that of the scientific inquirer, like Pythagoras, who asked, for example, "What are the reasons for the musical scale, and what are its limitations?" The first is the easy, laissez faire attitude; the second is a critical and scientific attitude which made no great progress until the beginning of the renaissance. Its first prominent organizer, Helmholtz, digested material accumulated from all sources, and made fundamental contributions through laboratory researches discussed in his epoch-making volume, Die Tonempfindungen (1862). [Sensations of tone.]
>
> Psychology as an experimental science had its beginning only seventy years ago, and in the first half of that period, showed no interest in music. Thus the scientific approach to the understanding and mastery of music is relatively new, and antiscientific musicians are still with us in large numbers.[4]

Anti-scientific musicians are, for the most part, particularly suspicious of reproduced music. They often

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[3] Carl E. Seashore, In Search of Beauty in Music (New York: The Ronald Press Company, 1947), p. v.

[4] Ibid., pp. 4-5.

feel that it (the music) has been tampered with and is not longer true to the original. They seem unwilling to admit that present discoveries in sound reproduction make it possible to hear a program reproduced at a great distance, with greater clarity, definition, presence, and with less background noise than a listener seated at the back of the auditorium at the actual performance. This would be particularly true if, for example, the auditorium were acoustically poor and a busy street just outside kept the ambient noise background at a high level. As Seashore says:

> In recent years the development of the Acoustical Society of America has brought about a revolution in musical thinking. Research in musical acoustics is being put on a rigidly scientific basis and is making great progress. There is an awakening interest in what is called musicology, the science of music. This science has many branches, one of the most active of which is the psychology of music.
>
> In view of this new demand for and the new possibilities of a scientific approach to music and scientific foundations for musical education and musical theory in preparation for the teaching and study of music, it is time to inquire, "*What can psychology do for music*?"[5]

Seashore then compares the physical terms and their psychological counterparts as outlined in Chapter I of this thesis. His comment is:

> We learn that the sound wave as the exclusive source of musical tones has only four basic variables: frequency, intensity, duration, and wave form. On the basis of that, it has been found that the musical

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[5] *Ibid.*, p. 4.

organism must have four corresponding capacities for hearing all music: the sense of pitch, the sense of loudness, the sense of time, and the sense of timbre. This conception simplifies the understanding of the nature and function of the musical mind in that each of these four basic functions appears in such complex musical forms as harmony, melody, dynamics, rhythm, volume, and tone quality. It has been shown that all of our musical memory, musical action, and musical composition may be expressed in these four terms. Thus the classification vastly simplifies the task of the musician and makes the problems of appreciation and performance concrete and specific.[6]

It is through this reasoning that Seashore feels musical esthetics can be discussed and analyzed using understandable and meaningful terminology. His remark is:

> Psychology organizes the scientific description of musical tones and the means for producing them. Psychology enables the musician to think in orderly, specific, describable, repeatable, and verifiable terms. All this is new to the traditional nonscientific musician. For example, he is interested in tone quality. But what is tone quality? What is its relation to other attributes of tone? What are its determinants? What are the limits, possibilities, and means for its mastery? Which, if any, of the accretions of scores of fantastic names for tone quality are significant, definable, and usable? These are all psychological questions with a musical meaning which may be taken into the laboratory.
>
> One element of tone quality is timbre, but, until recently, no music book revealed an adequate understanding of this concept. Definitions were often meaningless, and the waste of time and efficiency in teaching the mastery of timbre has been prodigious, largely because neither teacher nor pupil knew what was to be developed and had no objective standards for orientation.[7]

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6 Ibid., p. 5.

7 Ibid., p. 6.

Should the reader feel that this book lists all the answers to questions of musical esthetics, the writer hastens to add the following statement by Seashore:

> The scientific procedure in a new and unlimited field of this kind is a slow and arduous process, and in any generation mere beginnings can be made. But, as in the introduction of scientific methods in the classification of plants and animals and the interpretation of their complete life histories, once the scientific attitude is made possible, the purely speculative will gradually become less and less acceptable as a final solution. More progress toward a scientific approach to musical esthetics has been made in the last twenty years than in all preceding history.[8]

Much of the material in this book is openly experimental and incomplete. It is the scientific approach that is significant. It is hoped this approach will be followed throughout this study. Nothing can be gained by rejecting new or unfamiliar ideas before they are tried.

Many more books, pamphlets, monographs, and periodicals could be discussed as available literature, however, every important area relating to the body of the thesis has been included. This thesis topic has been pursued as a hobby for more than twelve years. In addition to the accumulation of literature, the writer has constructed many radios, recorders, and phonographs of all types in almost all price ranges from small shelf radios to $1,200.00 custom-made reproducers built to fit a particular home

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[8] Ibid., p. 15.

environment and the tastes of the owner.

The writer constructed a series of reproducers for the Stockton schools from 1946 to about 1952. The experimental nature of these machines and an evaluation of their usefulness appears in Chapter III.

The writer has given comparison demonstrations and lectures using poor quality and high-quality reproducers and conducted careful tests to learn the reactions of audiences and individuals, both musicians and laymen, to certain types of reproduced sound. Systematic construction and reconstruction and endless aural testing and evaluating has been carried on for years as the writer has attempted to resolve mechanical, psychological and esthetic problems to improve the reproduction and appreciation of recorded music. Many of these tests and their evaluations appear in Chapters III and IV.

It is the writer's hope that this thesis fills a need that has not been met by any other single study. The writer predicted twelve years ago that a demand for high-fidelity music reproduction would one day sweep the country. That day seems to have arrived and this study can at best be considered as only one of many such endeavors in the years to come.

It is hoped that the reader is now armed with an awareness of the many problems to be confronted, and some knowledge in acoustics, electronics, and music with which to understand more adequately and evaluate some of the proposed solutions to those problems.

CHAPTER III

EXPERIMENTAL TECHNIQUES AND SOME CONCLUSIONS REACHED IN CONSTRUCTING MUSIC REPRODUCERS

Chapters I and II of this thesis are written to guide the reader toward a better understanding of this chapter and Chapter IV. Chapter I serves to expose many of the problems that must be faced in the areas of acoustics, electronics, and music before adequate music reproduction can be brought about. Chapter I also lists some of the terminology common to these fields and actually defines many of the terms most often encountered in this study. Chapter II serves to (1) guide the reader toward useful sources of information of all kinds (2) provide adequate background material in each of the three areas--acoustics, electronics, and music, and (3) review representative studies from the bibliography to show some of their advantages and disadvantages interpreted in light of the subject of this thesis.

This chapter is planned to take the reader from this point of preparation through a step-by-step explanation and evaluation of most of the mechanical problems that must be solved before the adequate reproduction of music can be realized through electronic instruments. Psychological, psycho-acoustical, and esthetic considerations are, for the most part, deferred until Chapter IV.

In 1940, this writer purchased a new and up-to-date console model radio-phonograph manufactured by a well-known company. This machine was a revelation of tonal beauty in comparison to the inexpensive table model that preceded it. A full, rich bass was heard while the small radio seemed, by comparison, to have no deep bass tones at all. The music was more distinct and louder without being obtrusive, and instruments were identifiable in the reproduced sound that were literally not heard at all on the small table model.

What mechanical features did the new machine possess that the smaller one lacked? Could these features be improved upon still more through some study and even additional expenditures of money? The writer, spurred on by these stimulating possibilities, began to search for answers to these questions. A study of the circuit diagrams for each radio unit disclosed that the basic circuits were very similar even to the employment of some of the same tube types. The power output of the audio section of each radio was almost exactly the same. Certainly it was not in the electronic area that these two radios possessed any significant differences in construction detail. However, attaching the loudspeaker wires from the small radio to the large speaker on the new machine revealed a startling fact. The small radio sounded almost as good as the large one when its electrical output was channeled through the larger speaker. Moreover, the aural impression was that the larger speaker produced twice as much volume as the small speaker at the same volume control setting. In other words, the larger speaker was producing sound more efficiently and with more intensity than the smaller one using the same amount of input voltage.

This obvious difference meant that not only were the audio amplifiers not contributing any significant differences to the sound, but also the two phonograph pickups must be quite similar since records played on the two machines sounded quite alike in quality as long as the large speaker produced the sound. Examination showed that the two pickup cartridges (that part that holds the needle) were manufactured by the same company and were quite similar in appearance, tending to substantiate the fact that the loudspeaker was the unit most responsible for the obvious difference in quality between the two phonographs. At about this time (1941) a new and very powerful magnet was being used in the construction of loudspeakers. Current literature made much of the fact that this new magnet made the construction of better and more efficient loudspeakers possible, because of the fact that the frequency response and power handling capacity of a speaker was determined in proportion to the power of the magnet surrounding the voice coil. The following statement appears in a speaker catalogue:

> Now, with the advent of the new series of Alnico 5 speakers, we find it desirable to revise the system of model identification in keeping with a series of magnetic energy intervals. The magnetic energy is the energy in the air gap and is a measure of the acoustic output obtainable for a given electrical input to the voice coil.[1]

The fidelity of the speaker was supposed to improve also with the use of this stronger magnet. Note the following advertisement in this regard:

> MODEL 6201. A deluxe coaxial high fidelity speaker for the finest installation. Features built-in 2000 cycle crossover network and integral variable high frequency control. Extra heavy new University "W" magnet assures true response on lowest and highest notes. Frequency response 45-15,000 CPS. Handling power 25 watts.[2]

Examination of the magnets attached to the large and small speakers being tested disclosed an obvious difference in favor of the larger unit. Logic therefore pointed to the purchase of a new speaker with a larger magnet and greater power handling capacity. Further research disclosed that the larger paper cone of the big speaker produced more sound than the smaller diameter cone because its greater cone area displaced proportionately more air. A disadvantage became

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[1] Jensen Manufacturing Company, Fine Acoustic Equipment, Catalog No. 1010 (Chicago: Jensen Manufacturing Company, [n. d.]) p. 22.

[2] Walter Ashe Radio Company, Radio, Television and Electronic Equipment (St. Louis: Walter Ashe Radio Company, 1951) p. 27.

apparent in that larger speaker cones, while they produce low tones well due to the larger diaphragm movements, produce the high frequencies less efficiently, because of the uneven vibration of the more massive cone structure. As Beranek states:

> In the higher frequency range the cone no longer moves as a single unit, and the diaphragm mass M md and also the radiation impedance change. These changes may occur with great rapidity as a function of frequency. As a result, no tractable mathematical treatment is available by which the exact performance of a loudspeaker can be predicted in the higher frequency range.
>
> . . . .
>
> A logical means, therefore, for improving the high-frequency response would be to design the diaphragm so that at the high frequencies only the portion of it near the voice coil will move.
>
> . . . .
>
> Another means of accomplishing the equivalent of several sizes of cones is to mount two or more loudspeakers of different diameters near each other. An electrical network, called a crossover network, is used to supply electrical power to one loudspeaker at low frequencies and to the other, or others, at higher frequencies. . . . the loudspeakers are often mounted concentrically i.e., the smaller loudspeakers are placed in the front of and on the axis of the larger loudspeaker . . . . In the vicinity of the crossover frequency there is usually some shielding of the radiation from the larger loudspeakers by the smaller ones, with resulting irregularity in the response curve.[3]

This seemed to be the answer then, a two-way speaker "system"--one large speaker with a strong magnet to produce the low frequencies and a small cone speaker designed to produce just the high frequencies. This reasoning resulted in the purchase of a Jensen Model JCP-40 (twelve-inch

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[3] Beranek, op. cit., pp. 199, 201-202.

speaker). This equipment is described as:

> For FM-AM receivers and reproduction of commercial phonograph records where minimum space is an important factor. Excellent modernizing unit for replacement of single radiator, 12-inch speakers in radio receivers and phonographs. Can be mounted above the baffle for 10-inch speakers. Frequency Range: 50 to 12,000 cps. Maximum Input: 10 watts. Field: PM. Input Impedance: 6-8 ohms.[4]

An additional increase in efficiency and quality was at once noted by making "side-by-side" comparisons with the old 12-inch speaker. The Jensen Manufacturing Company in literature accompanying their speaker unit recommended the use of a bass-reflex type enclosure in which to mount the speaker and thus improve its low frequency response. The original cabinet for the large phonograph had an open back and did not seem to follow the design requirements for any particular baffle suggested in available literature. (See Chapter I (page 32) for types of baffles and descriptions.)

The following statement from a catalogue defines the reflex enclosure:

> The Bass Reflex principle, pioneered by JENSEN, is used in all JENSEN cabinets and reproducers to give maximum extension of l-f [low-frequency] response, free from objectionable "boom" or resonance. Bass Reflex cabinets employ an auxiliary port. Through exact acoustical proportionment of the port and cabinet volume, the port is made to become an auxiliary radiator at low frequencies, increasing efficiency by controlled utilization of energy which would otherwise be wasted.[5]

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4 Jensen, Catalog No. 1010, p. 5.

5 Ibid., p. 7.

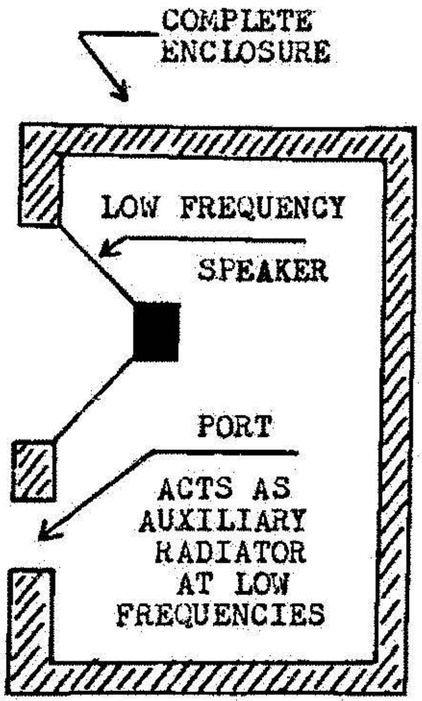

FIGURE 6*

BASS REFLEX ENCLOSURE

* Jensen, Catalog No. 1010, p. 7.

Dimensions were obtained for construction of this type of baffle from the Jensen Company. When this baffle was assembled and put into use, the additional low frequency response and "depth" added to the music was once more immediately apparent. It was difficult to make any kind of side-by-side comparison due to the time necessary to re-mount the coaxial speaker from one cabinet to the other. By the time the speaker was re-mounted, one could not be certain of retaining an accurate aural impression of the first test. This fact and the desire to "expand" the machine still further led to the purchase of a second Jensen JCP-40 twelve-inch coaxial speaker. Mounting one speaker in each cabinet with suitable switching arrangements made side-by-side comparisons on an almost instantaneous basis again possible. The efficiency and improved quality of either speaker operating in the bass reflex cabinet was beyond question of doubt superior to the old cabinet mounting.

During this testing, it was discovered that operating both speakers at once gave the reproduced music more breadth and richness through more spatial distribution of the sound throughout the room and the increase in air displacement due to the additional cone area of the second speaker. A second reflex enclosure was constructed and the two coaxial speakers were placed at the far end of the living room in opposite corners facing diagonally into the room. These positions were found to produce the best sound. Engineers at the Jensen Manufacturing Company conducted experiments in this matter which are described as follows:

> The corner position is definitely to be preferred, with an efficiency approximately 5 db higher than for the wall position. Position 2 (wall and floor intersection) is the next best with a gain of the order of 3 db. All three positions are definitely better in the extreme low frequency region than the outdoor response.[6]

A secondary problem presented itself with the use of two or more loudspeakers operating at the same time. They operated most efficiently when the cones vibrated "in phase" with each other. The next statement helps to illustrate phasing problems:

> Phasing is concerned with the utilization of two or more loud-speakers in such a way that the sound from any one speaker does not cancel the sound from other speakers, resulting in materially reduced sound output. This is an important consideration where speakers face the same direction. The connections to the voice coil, whether in series or parallel, must be made in such a manner that at any instant the diaphragms are in the same position. That is, all diaphragms are moving outward at any instant or moving inward at any instant.
>
> . . . .
>
> Phasing is of least importance where loudspeakers are pointing in opposite directions in an outdoor area. However, when installed indoors, as two speakers are brought closer together in a smaller angular relationship, the necessity for in-phase operation becomes increasingly important.[7]

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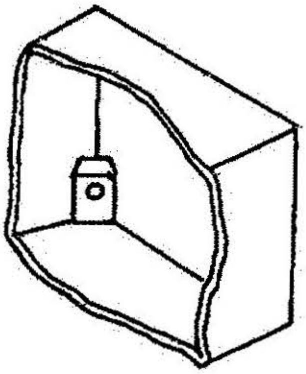
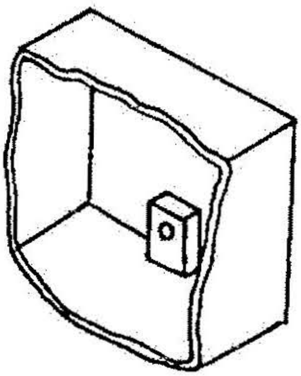
6 Jensen Radio Manufacturing Company, Technical Monograph Number One, "Loud Speaker Frequency-Response Measurements" (Chicago: Jensen Radio Manufacturing Company, 1944) p. 12.

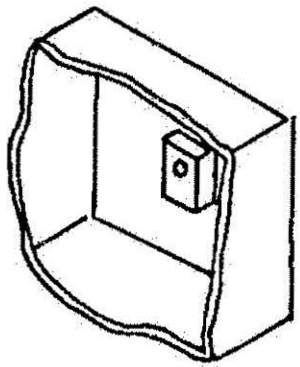
7 University Loudspeakers, Inc., University Speakers (White Plains, New York: University Loudspeakers, Inc., 1948) p. 19.



1. Corner Position



2. Wall and Floor Intersection Position



3. Wall Position

FIGURE 7*

ROOM PLACEMENT OF THE LOUDSPEAKER

* Jensen, Technical Monograph Number One, p. 11.

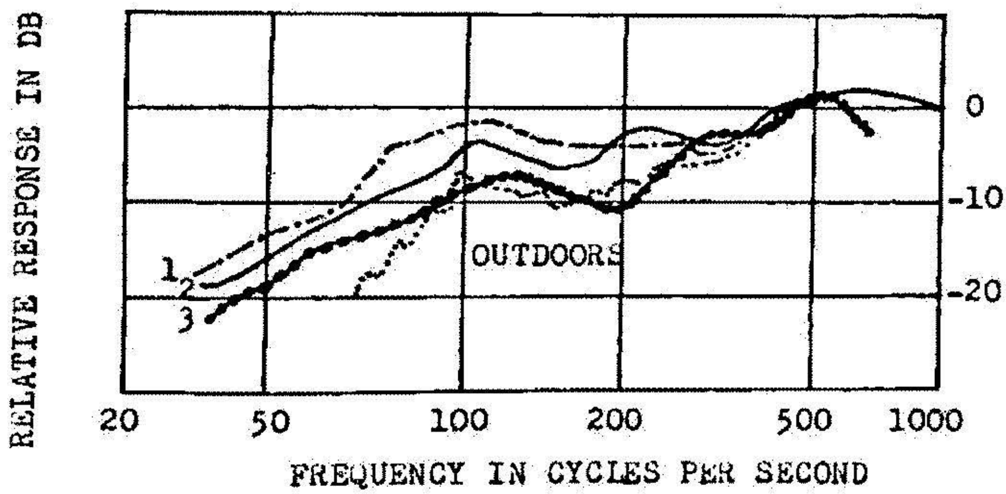


FIGURE 8*

LIVE ROOM RESPONSE FOR THE SPEAKER

IN DESIGNATED LOCATIONS

* Jensen, Technical Monograph Number One, p. 11.

Out-of-phase operation reduced the sound output disproportionately at different frequencies causing the musical balance to change unfavorably, particularly at low frequencies. A curious aural effect is obtained by walking slowly across the room when two speakers are operating out-of-phase. When your head passes the point where the sound reaches your ears at almost the same moment, cancellation takes place in the sound waves and the apparent volume level drops sharply, and you hear the two separate sound sources quite distinctly. If you keep moving, the effect disappears; but you are most conscious of hearing the speaker toward which you are moving.

A more reliable method of testing for phase relationship between speakers is to apply a moderately strong electrical signal to both speakers at once. By interrupting the signal with a switch and watching each speaker cone carefully, you can observe in which direction the cone jumps for the initial percussive click when the circuit is closed. If both cones move forward to produce the click, the speakers are operating in-phase. If one cone moves forward and the other back, the wires on one of the speakers should be reversed.

Another electrical consideration that is of prime importance is connecting new speakers to amplifier output terminals with proper impedance matching. Engineers at the

University Speakers Company illustrate the problem as follows:

> Sound system installations, whether of the simplest type involving a single amplifier delivering its entire power output to a single loudspeaker, or a group of amplifiers feeding unequal amounts of power to several hundreds of speakers, depend for satisfactory operation primarily on an efficient transfer of power from the amplifier to the speaker. If this transfer of power from source to load is to be accomplished with the maximum possible efficiency, it is essential that the impedance of the source be matched by an equal impedance presented by the load.
>
> For the highest possible fidelity, the source and load impedances should be matched within 10%. In practice, however, unavoidable mismatching up to 25% is tolerable, and acoustically the loss due to this mismatch is relatively small. Where a mismatch must occur, it is better always to connect to an amplifier impedance output tap lower than the load impedance to minimize loss of power and distortion.[8]

Not only must the voice coil impedances match with the speaker output impedance, but each speaker must have the same voice coil impedance if equal power is to be received by both speakers. The University catalogue goes on to say:

> When the voice coil impedances of speakers in series are alike, equal distribution of power will occur. However, if one speaker is 8 ohms and the other 16 ohms, the 16 ohm speaker will receive twice as much power as the 8 ohm speaker.
>
> In circuits employing a number of similar impedance speakers in parallel, each will receive the same amount of power. When a number of speakers of different impedances are paralleled, the lower impedance speaker will receive the greater amount of power. If one speaker is

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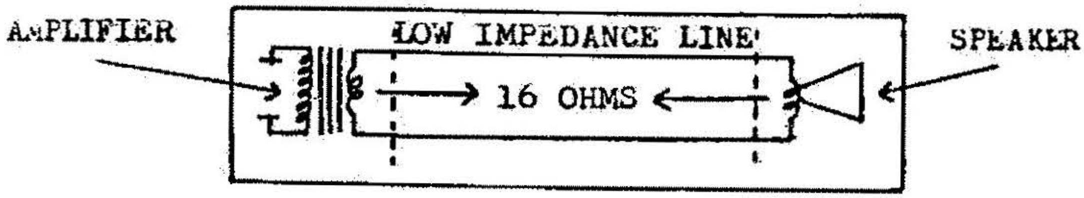
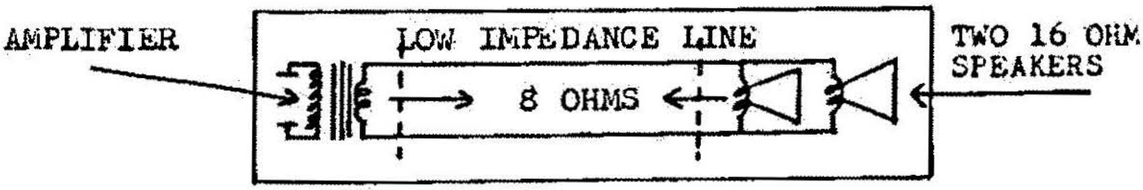
[8] *Ibid.*, p. 17.

In Fig. 1 below, which represents a typical simple paging system, the 16 ohm source impedance is properly matched by the 16 ohm load impedance presented by the speaker.



TWO OR MORE SPEAKERS IN PARALLEL. When the individual impedances of all speakers connected in parallel are the same, the resulting impedance is equal to the impedance of any one speaker divided by the number of speakers. In Fig. 2 below, the load impedance equals $\frac{16}{2} = 8$ ohms.



TWO OR MORE SPEAKERS IN SERIES. The total load impedance presented by several speakers in series is obtained by simply adding the individual impedances. In Fig. 3, the load equals 8 ohms + 8 ohms = 16 ohms. In most installations, series connection should be avoided since an open circuit in any one speaker will cause the entire system to become inoperative.

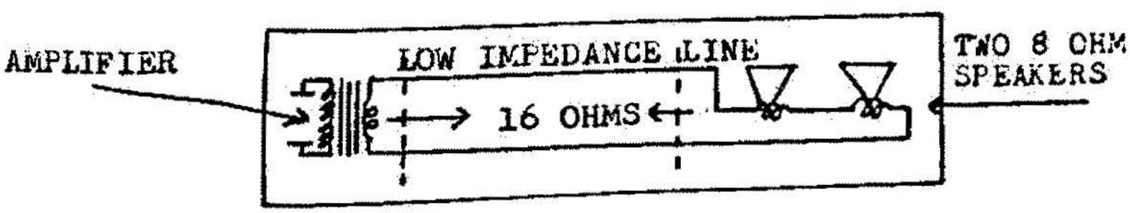


FIGURE 9*

SPEAKER MATCHING--LOW IMPEDANCE LINES

* University, op. cit., p. 17.

8 ohms and the other 4 ohms, twice the power will be received by the 4 ohm speaker.[9]

It now becomes clear that even with the same voice coil impedance, two speakers will only produce a similar amount and quality of sound if the magnet size, cone size, and relative efficiency of the speakers are the same, and if the same type of baffle enclosure is used. It is therefore recommended in multiple speaker systems that identical speaker units be purchased from the same manufacturer and each speaker enclosure should have identical dimensions and construction detail.

Two or more speakers properly matched electrically and acoustically and carefully placed in a listening room can give rich and somewhat "binaural" effect to the reproduced sound. (See Chapter V for a description of binaural sound equipment.) A listener seated at the far end of a listening room hears the sound from both speakers almost equally well. The blending of the two sound sources gives the aural impression that the sound source is actually located half-way between the two speakers in the center of the room. In listening tests involving different kinds of music, very critical listeners perceived some musical instruments in the reproduced sound that seemed to be

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9 Ibid., p. 18.

coming from either one speaker or the other and not from the blending of the two sources at the center of the room. This was not a consistent impression in that the same instrument in a different register (higher or lower in pitch) would suddenly seem to shift to the opposite speaker source. Investigation showed these aural impressions were due to resonant peaks in the individual speakers and enclosures. Since no two loud speakers and enclosures could be exactly alike even in terms of the density, strength, and flexibility of the wood, paper, and other materials used in construction, certain vibratory frequencies would of necessity resonate more efficiently in one speaker than the same frequency might in the other speaker. For example, if a violin sounded a tone at or near one of these resonant frequencies, that tone would be produced with the most intensity by that speaker and the listener would perceive that sound as emanating from that particular side of the room. A moment or two later the same violin might produce a tone at or near a resonant frequency of the opposite speaker and the perceived sound source would shift in that direction. Musicians used to orchestral instruments seated in a conventional pattern that does not change during performance often find these aural impressions interesting but musically somewhat disconcerting when, for example, the concertmaster of the violin section seems to have suddenly moved across the room during his rendition of a solo passage with the orchestra.

It seemed logical to assume that a single coaxial speaker with an extra-large low frequency cone could retain the advantages of sound produced by a large diaphragm area and eliminate the resonance problem just described. Listening tests also seemed to indicate that the small three-inch cone "tweeter" speakers contributed noise and rattle when extremely loud passages were reproduced. A large tweeter speaker with greater power handling capacity was therefore made an additional requirement for the new unit. With these requirements in mind, a Jensen Model JAP-60 fifteen-inch speaker was purchased. A catalogue description of this unit follows:

> For FM-AM receivers, high quality phonographs, reproduction of commercial phonograph records and similar applications where smoother response and better balanced characteristics are required than offered by Models JHP-52 or single, direct-radiator speakers. O.D. 15-1/8"; Depth 8-1/8"; Baffle opening 13-3/4". Frequency Range: 50 to 15,000 cps. Maximum Input: 17 watts. Field: PM. Input Impedance: 500-600 ohms. Network: Integral two-channel type. H-F Control: "Roll-off" type with 4-position switch.[10]

A new bass reflex cabinet designed for the larger fifteen-inch cone was constructed and in side-by-side comparison tests, the new unit out-performed the two twelve-inch

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10 Jensen, Catalog No. 1010, p. 5.

units in both efficiency and quality. The definition of the sound was better and the high-frequencies were "crisper" giving the reproduced sound more "presence." Although the larger tweeter (five-inch) on the new speaker was better than the two three-inch tweeters, there was still a noticeable noise contribution from the tweeter during loud passages. The noise was almost eliminated by turning down the output level of the tweeter, but this also reduced the production of the high frequencies in the music. This solved the noise problem but at the cost of reducing the presence and fidelity of the sound. Realistically, this could not be accepted as a solution if true high frequency reproduction was to remain as a necessary pre-requisite for high-fidelity sound.

In 1943 a friend living in the East wrote a letter indicating his imminent induction into the armed services and listing his various "Hi-Fi" components that were to be sold. Among these items was included an excellent eighteen-inch auditorium speaker and a horn-type tweeter employing a small metal diaphragm to produce the sound in place of the paper cone. (See Chapter I, page 35 for horn-type speakers.) Horn-type loudspeakers are substantially more efficient than cone-type direct-radiator loudspeakers and a possible solution to the tweeter noise problem seemed likely. In due course of time, both units were purchased. The eighteen-inch auditorium speaker came mounted in a well-designed reflex baffle and the horn-type tweeter was placed on top of the enclosure to associate it aurally as closely as possible with the larger speaker. The auditorium speaker was fitted with a new cone designed especially for low frequency reproduction. A special filter unit permitted only appropriate high frequencies to pass to the tweeter voice coil, thus completing a full range 40 to 15,000 cps. speaker system. (See Chapter I, p. 36 for full-range reproduction requirements.) Side-by-side listening tests were then conducted with the fifteen-inch coaxial speaker unit. It is unfortunate that the reader could not at this moment hear such a side-by-side comparison under controlled listening conditions. It is difficult to describe the difference in words--it needs to be heard to be fully appreciated and understood. The new system seemed to double the volume for a given input voltage. The bass tones, such as organ pedal stops and bass drum beats, were so realistic in comparison to the fifteen-inch speaker that listeners openly exclaimed at the difference.

The horn-type tweeter produced the high frequencies so clearly and crisply that triangle, snare drum, and tamborine parts were heard in orchestra recordings that had never been perceived by dozens of previous hearings. It was during this period that the writer gave weekly record concerts as a member of the Armed Forces. Each concert was followed by a brief discussion and demonstration of the equipment to those interested. There was never a lack of listeners, and friendships developed that gave stability and a more critical atmosphere to the aural evaluations. Scarcely a month went by without some significant change in the mechanical or electrical portions of the phonograph.

Other changes were taking place along with the evolution of the speaker system already described. The original console phonograph would never produce enough undistorted volume even with more efficient speakers to keep the full attention of an average audience on the music and produce the full climaxes of the orchestra with realism. A special fourteen watt amplifier was purchased from the Bogen Manufacturing Company, Model PV-10, built especially for phonograph reproduction. It was a resistance coupled amplifier with a wide-range frequency response and a built-in circuit called "Volume Expansion." (See Chapter I, page 24 for volume expanders.) The extra power and absence of audible distortion made this amplifier seem most adequate and strangely enough, although other amplifiers were tested from time to time, no basic change was ever made in the amplifier itself through more than twelve years of experimentation. Some justification for this fact is found in the following quotation:

> Laboratory tests and carefully controlled listening tests on thirty-six amplifiers, ranging in price from $41.75 to $347.70, indicate that there are many good amplifiers on the market, with the most expensive models tested offering no advantage over some amplifiers selling at very much lower prices--no advantage, at least, that the highly trained ears in CU's listening jury could detect. The highest rated of the amplifiers tested sells for $99, and the second best--an outstanding best buy--costs only $69.50. . . . CU's electronics engineers believe that money saved by the purchase of a relatively inexpensive amplifier can be spent to much better advantage on other components of the high-fidelity reproducing system, such as the speaker and enclosure, the pickup and turntable, or the tuner. Additional expenditures on speaker and enclosure are most likely to result in major improvement in tone quality.[11]

The same day that the new amplifier was incorporated into the radio-phonograph, the phonograph pickup was placed on a record with the turntable stopped. The volume control was turned quite high and there resulted an immediate and painful roar from the loudspeaker. The roaring stopped only when the volume was turned very low or the pickup was taken off the record. This never had happened with the old radio amplifier because of its lack of power. This objectionable sound was the phonograph's equivalent of the howl heard on a public-address system when the volume is turned too high. The effect is called "feedback." When the phonograph pickup was placed on the record the needle moved enough to move the attached record turntable enough to move again the

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[11] "Hi-Fi Amplifiers," Consumer Reports, 19:154, April, 1954.

needle and re-amplify the already amplified sound. These cycles of vibration and re-amplification occured over and over so rapidly that a roar of sound was the result.

The turntable and pickup were mounted on free-floating springs and little more could be done to isolate the cabinet vibration from the turntable. Yet when the turntable and pickup units were held just out of contact with the cabinet, the feedback stopped, only to start again when even one corner of the unit touched the cabinet. The answer was to separate the turntable and loudspeaker units as much as possible. Placing the loudspeaker cabinet at the far end of the room not only reduced distortion caused by feedback, but permitted the listener to adjust records, volume, tone, etc., without the necessity of moving from his chair. All these units could be mounted, of course, at any convenient point, as far as necessary from the acoustical sound source. It is difficult to understand why commercial radios and phonographs have not incorporated this feature as a selling point in new model designs. It was found helpful to mount the speaker enclosure on rubber cushions to reduce floor vibration transmitted from the cabinet. A carpet placed in front of the speaker cabinet also assisted in reducing the reflection of sound waves off the floor that tended to cause "boominess" at very low frequencies.

The operation of the first extended range coaxial speaker with the phonograph brought out the fact that the phonograph pickup was not transmitting even reasonably good frequency response from the record to the amplifier and speaker. Purchase of an expensive light-weight arm and a wide-range crystal cartridge resulted in an immediate improvement in over-all quality and the small tweeter speaker began emitting a steady "hiss" that was to become the trademark of high-fidelity reproduction on the old 78 RPM records. Record manufacturers included abrasives such as carborundum in their records to help the records wear longer. The fine particles of abrasive transmitted their presence to the listener through an audible hiss as the needle passed through the groove. Unfortunately the frequencies of the scratch and the frequencies of many of the overtones in the music were similar enough to be inseparable. Therefore, high-fidelity reproduction was possible with "needle scratch," or the needle scratch and the high-fidelity could be eliminated by turning down the treble control or turning off the tweeter and eliminating most of the high frequencies. Many ingenious solutions to this problem were discussed in current periodicals and monographs up to the advent of the long-playing record which was manufactured with little or no abrasive content. One of the least objectionable scratch suppressors is explained in the following remarks by A. C. Shaney:

In attempting to find methods for suppressing scratch, the following 2 solutions presented themselves:

(1) Inasmuch as scratch is a conglomerate of indiscriminate frequencies, and music is characterized by discrete frequencies, it seemed possible to develop a set of discrete and indiscrete filters, each feeding into control-grids of a differential amplifier arranged so that when a preponderunce of indiscrete frequencies were present, the volume would drop, and when a preponderance of discrete frequencies were present, the volume would maintain its average level. This effect would ultimately produce lower levels when scratch was present and normal levels when definite signals were being reproduced.

(2) Scratch is not objectionable at high-level outputs, but gradually becomes more and more objectionable as the signal level decreases, only because the signal-to-scratch ratio has de-creased. It follows than an effective method of decreasing objectionable scratch is to automatically lower the overall gain of the amplifier at low levels and maintain its average level at normal levels.

The latter method seemed to offer a simpler solution. In the process of the development of this circuit, it was found that the functions of both the expander and the suppressor could be combined in the same pair of tubes, it being only necessary to increase the level above predetermined average for expansion and to decrease it for scratch suppression.

It was further found, that the speed of scratch suppression needed adjustments for different selections, because of the fact that the sensitivity of the ear does not change instantaneously with changes in level. To provide a wide degree of time delay control, a full-wave rectifier is employed, which couples to a separate push-pull control voltage amplifier. The use of full-wave rectification (which doubles the frequency of the rectified voltage) provides for small capacitative filters which thereby enable the use of high-speed control circuits without introducing hash into the signal control-grids of the expander-scratch-suppressor circuit. The degree of expansion and suppression is controlled by the "Expressor" control (which is a contractual abbreviation for expander-suppressor).[12]

Any other form of scratch suppression results in a reduction of fidelity and therefore is not really an acceptable solution. It must be said however that many people prefer limited fidelity with little or no scratch. This preference usually is based on past conditioning through commercial radios. This type of problem is discussed in Chapter IV.

As a mechanical device, phonograph pickups seem to be one of the least efficient units in the long chain of devices that reproduce music, reaching from the microphone to the loudspeaker. (See list of pickup design problems in Chapter I, page 27). A light-weight arm with low needle pressure (one ounce or less) and some type of permanent needle usually give the best quality and wear the records the least. Diamond or sapphire needles have worked best in that they retain their tip radius dimensions the longest. Iron needles wear the quickest and if they are not changed often will seriously damage the fidelity of a record by gouging out the most delicate portions of the record groove, which are the high frequency vibrations.

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12 A. C. Shaney, Amplifier Manual (New York: The Amplifier Company of America, 1941), p. 15.

The relatively new magnetic cartridge has proved to be more stable than the crystal cartridge in that it does not fail because of excessive heat as does the crystal, and it has certain mechanical advantages. This advertisement describes the stylus and vibrating parts:

> It is truly one of the remarkable achievements of the electronics industry. The pickup has a permanent natural sapphire stylus having an included angle of 45° to 50° and a tip radius of .003" ± .0002". The small mass of the moving system provides a very low mechanical impedance resulting in a great reduction in record wear and reduction of audible "needle chatter" to the near-vanishing point. Reduction of annoying needle chatter is of importance when the reproducer is operated in the same room with the loudspeaker.[13]

All pickups have an uneven frequency response and need some type of "tone equalizer" to help make their output equal at all frequencies. Condenser and resistor networks are common for crystal pickups, such as the Astatic E4P tone equalizer. The E4P is described as follows by the manufacturer:

> Model E4P is an adjustable tone compensation network to be connected between a crystal phonograph reproducer and the volume control of the amplifier (see circuit diagram) to vary the response characteristics to suit individual taste and circumstances. A rotary switch controls the different equalizing circuits which are as follows: (see graph)
>
> 1. BASS-Lows accentuated.
> 2. MEDIUM-Lows predominate.

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13 Electronic Ideas Incorporated, The New 3-A Lateral Reproducer Assembly (California: Electronic Ideas Incorporated, [n. d.]), p. 2.

3. HI-FIDELITY-Commercially cut records practically flat.
4. OUT-ALL equalization eliminated giving normal response of a crystal pickup.

Equalization is obtained with relatively small loss of power; the level is lowered on an average of only 4 db. throughout the useful frequency range except where high frequencies are purposely attentuated (see graph).[14]

The disadvantage in this type of equalizer is that the various switch positions available make an abrupt change in the quality of the music with no "in-between" settings possible. Magnetic reluctance type cartridges have a very low voltage output. Standard practice is to include tone compensation in a pre-amplifier which is designed to correct the pickup frequency response and build up the output voltage enough to connect to the standard phonograph input on an amplifier. More elaborate equalizers are available at a higher cost, such as the 20-C switch assembly. The manufacturer's description follows:

The 20-C switch assembly is the frequency response control unit to be used with the 20-B compensator. The switch has five positions that cover the great majority of today's requirements in transcription and phonograph reproduction. The switch positions and their explanations are as follows.

ORTHO--"Ortho" is an abbreviation of the word Orthacoustic used by the Radio Corporation of America and designates the recording frequency characteristic

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[14] Astatic Microphone Laboratory, Inc., Data Sheet No. 107 (Youngstown, Ohio: Astatic Microphone Laboratory, Inc., n. d.), p. 1.

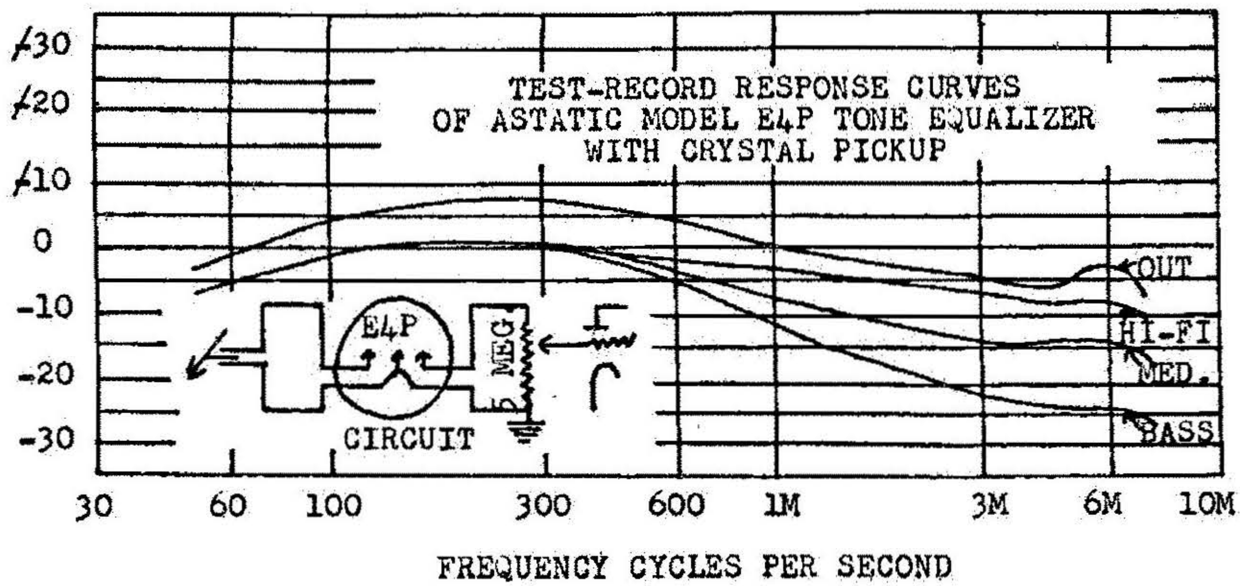

FIGURE 10*

TEST-RECORD RESPONSE CURVES

* Astatic, Data Sheet No. 107, p. 1.

used by that concern on its transcriptions. This standard recording characteristic has also been adopted by the National Association of Broadcasters (N.A.B.) and is used by most large recording companies manufacturing transcriptions for broadcast purposes. A transcription recorded according to this standard and reproduced with a 3-A reproducer set on the "ORTHO" position will result in an overall response flat from 40 to 10,000 cycles per second.

FLAT--In this position the 3-A system is flat from 40 to 10,000 cycles on the basis of 500 cycle cross-over frequency. Records recorded with the standard 500 cycle cross-over are recorded on a "constant velocity" basis above 500 cycles and a "constant amplitude" basis below 500 cycles. Inasmuch as magnetic devices are inherently constant velocity in nature, a 6 db. per octave boost below 500 cycles has been incorporated in the 20-B compensator to allow flat reproduction of standard records over the range mentioned. This position may be used for laboratory work, transcriptions recorded "flat," or phonograph records of the noiseless type (Vinylite).

1.--This position inserts a cut-off filter that attenuates all frequencies above 5,000 cycles at a rate of 8 db. per octave. Position 1, should be used on good shellac records to cut off "top end hiss" without noticeably affecting quality of reproduction.

2.--This position inserts a cut-off filter that attenuates all frequencies above 3,000 cycles at a rate of 8 db. per octave. Position 2. should be used on noisy and worn records to reduce scratch.

3.--This position inserts a cut-off filter that attenuates all frequencies above 1,500 cycles at a rate of 8 db. per octave. Position 3. should be used for badly worn or very old records to reduce scratch and noise. This position is also of great value when used with sound effects records where surface noise is frequently a serious problem.[15]

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15 Electronic Ideas, <u>op</u>. <u>cit</u>., pp. 2-3.

Since good pickup performance is most essential to high quality music reproduction, time and money spent on a carefully chosen unit will reward the spender many times over. (See quotation from <u>Consumer Reports</u> on page 76 of this chapter.)

Closely aligned with the problems of pickup performance are those associated with the record turntable and its task of revolving the record at a fixed and constant speed. Mr. Ted Powell, electronics expert, describes these problems as follows:

> Several distortion effects are developed by the turntable assembly itself. Some of these are not generally appreciated.
>
> Vibration in the motor due to gears, bearings and A.C. introduce noise frequencies and their harmonics. These effects are noticeable as turntable "rumble," power-frequency hum or mechanical groaning or grinding noises. Turntable rumble can be heard even on some costly transcription type phono assemblies. It is difficult to eliminate with turntable assemblies designed as they are, especially with those using ball-bearings and gear drive. This trouble is still further aggravated in hi-fidelity amplifiers with genuine and not synthetic low-frequency response in the phono system.
>
> . . . .
>
> The other and more commonly recognized distortion effect developed by the turntable assembly is the one known as "wow." This is frequency-modulation distortion and is caused by variation of drive-motor speed. This speed variation is due to the variable turntable load which is caused by gear-tooth torque pulses and the variable modulation groove, the wider the stylus tip is swung, the more work has to be done to swing the stylus against the impedance of its mechanism. Therefore the transient load placed upon the turntable drive motor is greater. The result is

the familiar and inevitable "wow" where the usual turntable assembly is employed.[16]

The finest pickup can be reduced to ineffectiveness if the turntable does not accomplish its job smoothly and quietly. No less than seven different turntable motors were tried in the console radio-phonograph described as the test machine in this chapter.

More adequate loudspeakers, amplifiers, and pickups incorporated into the test machine helped to make the inadequacies of the record turntable more apparent. Many records when played on the turntable were reproduced at the wrong pitch level. Careful tests using a tuning fork as a pitch reference showed the pitch of music known to be in certain keys was being raised, or sharped, by the too rapid revolutions of the turntable. A new turntable with a device for adjusting the speed of the record revolutions was therefore required. Musicians with sensitive pitch perception are often annoyed greatly by music reproduced at the wrong pitch level. Such a control would permit accurate adjustment of all kinds of records to a prescribed pitch level and where pitch was not important, the music could be speeded up or slowed down, if necessary. This is often done with phonographs in dance studios.

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[16] Ted Powell, "Pick-Up Performance," Part II, Radio-Craft, 15:27, October, 1943.

Piling small lead weights on top of the record as it was playing seemed to reduce "wows" in the reproduced sound and improved the clarity of the bass instrument tones. The "wows" were reduced due to the flywheel effect of the weights revolving on the records, and the bass tone was improved through the elimination of lateral record oscillation by the locking effect of the weights. Mr. Powell has made these recommendations to manufacturers:

SUGGESTIONS TO MANUFACTURERS

. . . .

14.--Design turntables with massive rims in order to obtain more flywheel effect to reduce wow effects.

. . . .

16.--Make the turntable center-pin with a felt-padded lock-nut or else a split-expansion type center-pin with which to anchor down recordings on transcription type turntables to eliminate record disc lateral oscillation and radial slip and consequent small low-frequency cut-off and distortion effects.[17]

Some turntables induced hum into the pickup which was picked up and amplified through the speaker system. Any extraneous noise or hum adds to the general noise and raises the masking level of the ambient noise background. (See Chapter I, page 3 for sample masking levels, etc.) This hum was noticed to increase and decrease as the pickup was passed back and forth just above the rotating turntable with no record placed upon it. Thoroughly grounding the motor frame and shielding and grounding the pickup arm

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17 *Ibid.*, p. 55.

cartridge, and the wires leading to the amplifier usually eliminated all hum from this source.

Eliminating hum, tube, and other operating noises from phonographs in order to reduce background noise was a major problem on the test phonograph. Rim driven turntables and their associated driving motors contributed noticeable noise until they were encased in a box lined with heavy felt to absorb and enclose the sound. Transformer windings often hummed audibly as current flowed through them. This type of hum, while it is not loud, is most obvious when the equipment is first turned on before the tubes have warmed up and contributed their noise through the speaker. Isolation and sound absorbing materials wrapped around the offending transformers was a partial solution. A check can be made on the amount of noise contributed through the loudspeaker units by placing your ear against the speaker enclosure as someone turns on the amplifier and other equipment. As the power supply begins to operate, any inherent hum in its operation will be immediately apparent as a sixty cycle tone in the speaker. A moment later the tubes will warm up and any other sources of hum such as that caused by inadequate shielding will be added to the power supply hum (if any). The last sounds added to the speaker background noise are usually the popping and hissing of the tubes, resistors, condensers, bad connections, etc. Power supply hum usually can be reduced through replacing old or faulty filter condensers and possibly de-coupling certain stages within the amplifier.[18]

Check for other hum by plugging and un-plugging the pickup with the volume control wide open. If hum is increased by plugging in the pickup wire, that portion of the phonograph needs attention and additional shielding or grounding. If the hum is within the amplifier, attention by an expert repairman or the factory is usually warranted. Condenser and resistor noises are difficult to locate unless there is an actual part failure. Replacement is usually the only cure. Tube noises sometimes can be reduced by substituting various tubes until the quietest combination is found. Tube hiss can be reduced by not operating the amplifier gain at too high a level. One amplifier tested was designed so that the power output stage always operated at its maximum level and the input voltage was varied to adjust volume. Tube noises were very prominent and impossible to eliminate so the amplifier was soon discarded.

By careful and persistent efforts the hum and noise level in the test machine were reduced to a point where, with the unit on and all controls turned to maximum with no

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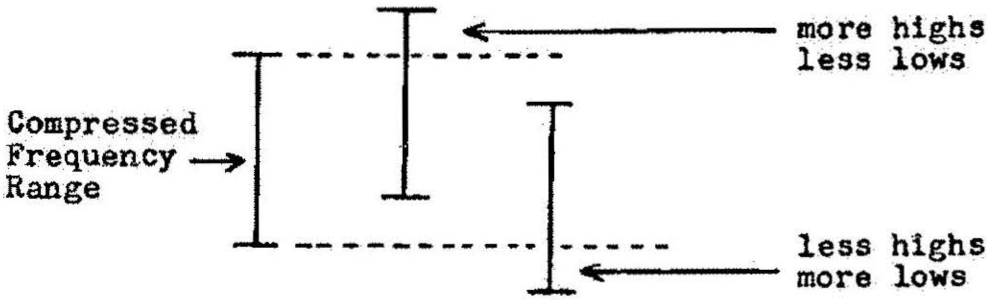
[18] See The Radiotron Designer's Handbook for decoupling procedures.

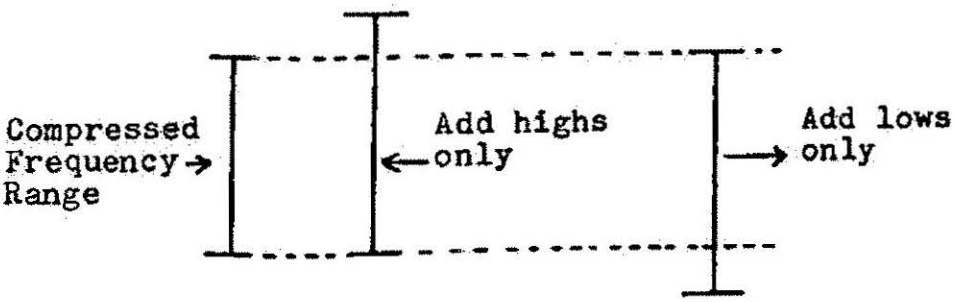
record playing, critical listeners entering the room would not be aware that the phonograph was on until they were asked to listen for and identify every noise they could perceive within the room. They would usually hear tube hiss first and then sixty cycle hum. With the controls set at normal listening levels, most listeners perceived no sound from the phonograph of any kind. Periodic attention had to be given to noise problems as various parts deteriorated with age.

The original radio-phonograph and the new fourteen watt amplifier were each equipped with only one tone control. This control was the attenuating type found on the majority of audio amplifiers. When this control is turned to its maximum fidelity position, the amplifier produces its full frequency range. When the control is turned counterclockwise, the high frequency response is reduced on a sloping curve similar to that found on the figure illustrating the response for the bass setting for the E4P tone equalizer, (see page 83). More bass is added as the gain is advanced to replace the volume lost on the highs. With this type of control it is impossible to amplify the low frequencies alone without reducing the highs and vice-versa. Mr. Stokowski in his book entitled Music for All of Us, has illustrated this problem and its solution very clearly. He says:

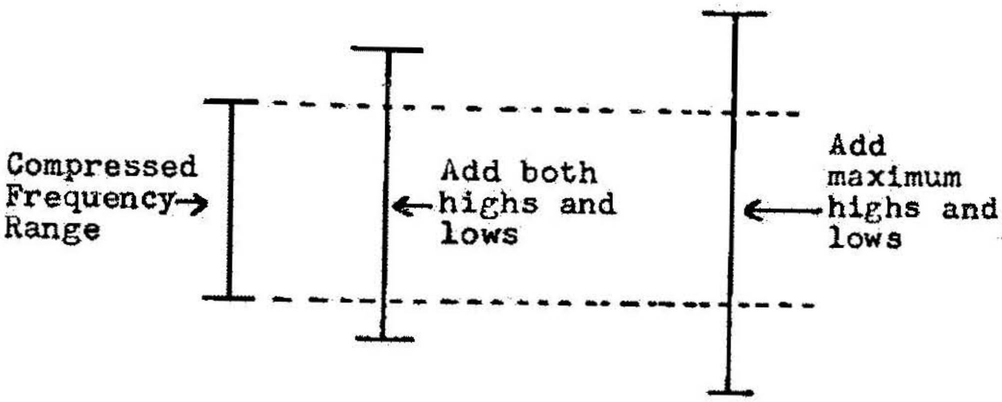
Every sound reproducer in the home, both for records and radio, should have independent control of high and low frequencies. When there is only a single control and we add intensity to the high tones, we do so at the expense of the low tones--and conversely, when we add depth to the low tones we do so at the expense of the high tones, like this--



If we have independent control over both high and low tones, we can increase the intensity of the highs without lessening the fullness of tone of the lows--or we can increase the intensity of the lows without disturbing the high sounds if they are already satisfactory, like this--



Or we can increase the intensity of both the highs and the lows to the degree that makes the music sound best, like this--



[19]

The solution to this problem was found through the construction of an electronic tone equalizer that actually amplified the bass and treble separately--the bass frequencies through an electronic circuit and the treble through an attenuator control similar to that described above. These controls, in conjunction with the volume control on the tweeter and the E4P tone equalizer with its various settings, gave an almost infinite number of tone control possibilities. Even records with poor tonal balance and poor fidelity could often be made to sound reasonably musical through careful use of the tone controls. For

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19 Stokowski, <u>op</u>. <u>cit</u>., pp. 254-55.

example, if a recording had excessive surface noise and lacked richness, the following procedure usually helped to a marked degree in improving the reproduction. Set the E4P pickup equalizer on medium fidelity, turn the electronic tone equalizer bass up to add depth and richness, and turn up the treble control to amplify those high frequencies that are getting through the E4P (usually those below the scratch frequencies). The high treble setting gives a pseudo brilliance that is often mistaken by the average listener for high-fidelity. The advanced bass setting gives added depth to the sound and a fairly high total listening level gives some impression of realism and presence. Adequate separate tone controls and preferably a volume control for the tweeter speaker were made requisites for future phonograph construction on the basis of this type of test and observation. The sensitive use of tone controls and volume settings are as essential to the level and balance of the reproduced music as the conductor's sensitive use of control of level and balance was at the original performance.

Before turning to the section on construction of reproducers for the schools, it seems advisable to summarize briefly the basic conclusions drawn in this chapter up to this point. These conclusions guided the planning of the school reproducers.

I. The speaker is perhaps the most important single unit contributing to tone quality and the definition of the reproduced sound.

A. The speaker must be housed in an adequate enclosure designed to certain specifications. These specifications change with speaker size and baffle types.

B. The type of enclosure should be chosen from known and accepted basic designs, such as the bass reflex and infinite enclosures previously described.

II. The larger the speaker cone and the larger its associated magnet, the more efficiently and clearly the speaker will reproduce a given input, both in volume and transient frequency response.

A. Large speakers give greater sound coverage through the larger displacement of air created by the increased cone area.

B. While large diaphragm speakers produce low frequencies very efficiently, they do not produce high frequencies clearly due to the irregular vibration patterns of the cone at high frequencies. (See previous explanation.)

III. The extreme high frequencies are produced best by a horn-type tweeter (metal diaphragm).

A. The horn tweeter is more efficient than cone-type tweeters and produces clearer highs. Cone-type tweeters do not as a rule have sufficient power handling capacity.

B. Dividing networks are necessary to provide the large and small speakers with only those frequencies they are designed to produce.

1. The tweeter speaker should be equipped with a volume control.

2. The horn-type tweeter should be placed close to the large speaker for best results.

C. Multiple speaker systems must be phased properly and reflect correct impedance matching for even power distribution and good fidelity.

IV. Speaker units must be placed carefully in the listening room to take advantage of the best acoustical sound reflection patterns.

A. Corner placement is best.

B. Wall mounting is poorest.

V. Adequate amplifier power is needed to produce the loudest passages clearly without distortion.

A. Volume expansion is an electronic circuit designed to help replace the volume lost through compression at the recording studio. With careful use, it can enhance the sound of the phonograph.

VI. Feedback and other associated problems are solved by physically separating the turntable and pickup unit from the speaker system.

A. Separate units permit more convenient operating procedures (i.e., changing records).

B. Separate units permit selection of a more adequate listening position by the operator.

VII. The best crystal pickups are adequate, but they are being replaced by magnetic units that are more efficient.

A. Diamond and sapphire stylii are best due to long wearing qualities.

B. Care of stylii is of utmost importance both to the life of the record and high-fidelity reproduction.

VIII. Scratch suppressors should not attenuate the high frequencies. The best scratch suppressor works through a kind of inverted volume expansion circuit. Volume expanders tend to reduce audible scratch because soft passages are in effect reproduced softer than expanded passages.

IX. Pickup equalizers are necessary to all types of phonograph cartridges.

A. They equalize normally uneven frequency response characteristics.

B. They permit control of the frequency response for various kinds of records, etc.

C. They act as an auxiliary tone control.

X. Weighted turntables reduce "wow" and other kinds of distortion.

A. Records should be locked on the turntable to reduce distortion and improve bass response.

B. Speed adjustment is essential for pitch control.

XI. The ambient noise background can be reduced through adequate shielding, filtering, and careful operation techniques.

A. Hum and noise reduction is a continuous problem.

B. No noise should be heard from the equipment at normal operating levels with the turntable stopped.

XII. Separate treble and bass tone controls are essential to good music reproduction.

A. Each level should be controlled separately without interaction.

B. Tweeter volume control is also a type of tone control.

C. Pickup equalizer is a type of tone control.

   1. The careful use of all tone controls gives an almost infinite number of tone settings.

   2. The tone of old records can be improved with careful tone and equalizer adjustment.

D. Different acoustical environments surrounding the reproducer demand different tone control settings to balance the reproduced sound due to the varied patterns of reflection and absorbtion from the walls, floor, and ceiling.

This outline presents most of the general conclusions that were reached on a more or less mechanical level after about eight years of study, experimentation, and aural evaluation. The aural test was the final and conclusive test. Tubes that tested as satisfactory on tube testers often were discarded after an aural evaluation of the tube in actual operation. Excessive hissing, popping, and microphone tendencies made many new tubes unacceptable. New and old condensers occasionally contributed noise or reduced quality and had to be replaced when electrical tests seemed to indicate they were still in good condition. Pickup cartridges and styli were replaced automatically every six to ten months because side-by-side tests almost always indicated that new ones improved the quality and clarity of the music. Amplifier output tubes were changed at least once a year due to decreasing efficiency and new tubes were matched carefully on a tube tester before installation. Occasional systematic checks of the equipment were made using known test records as comparison references to insure peak performance efficiency at all times.

Upon being discharged from the Armed Services after World War II and returning to college, the writer had occasion to observe classroom music teaching in the Stockton elementary schools. The radio and record reproducers used by the instructors were hopelessly inadequate and very old. Symphony orchestra music tended to sound like poor band music, with little or no recognizable quality to the sound of the various instruments. These school children obviously were being exposed to very poor tone quality and seemingly accepting it as representative of the things they were being taught. The writer proposed a solution to this problem based on the information accumulated from the earlier experiments already described in this chapter. "Why not build radio, phonograph, and public address reproducers for the schools using inexpensive equipment wherever possible, <u>but incorporating into the machines all those features found</u>

<u>previously to contribute to good tone quality and operating efficiency?"</u> Cabinets and speaker enclosures could be constructed by school carpenters. Parts could be purchased at a substantial discount through the schools, and each machine could be designed to include just those features considered essential for a particular class.

The coordinator of music for the Stockton schools approved the idea and plans were drawn up and put into operation. An upper limit for the cost of each completed reproducer was established at $135.00. One high school desired a machine for only 78 RPM record reproduction in a music classroom. An elementary school wanted a unit that would play records, both standard 78 RPM and 33-1/3 RPM transcriptions, provide for radio amplification, be mobile, and serve as a public address system in the school auditorium. Out of some twelve machines eventually constructed, only two or three were identical in every detail and yet they all incorporated the following basic requirements:

1. A large and powerful high-fidelity speaker mounted in a carefully designed bass reflex enclosure. (The speaker unit was placed in a corner at floor or ceiling levels wherever possible.)

2. An amplifier with reasonably good fidelity and plenty of power.

3. An adequate tone control system including separate

treble and bass controls and pickup equalization. (Since the inexpensive amplifiers selected did not have separate tone controls, a special tone control unit was built involving the addition of a tube and a small extra chassis.)

4. An inexpensive but light-weight pickup arm with a permanent sapphire stylus and a high-fidelity crystal cartridge.

5. A separate pickup and turntable cabinet, using an inexpensive turntable motor, rim driven for more stable operation.

6. Some electrical access plugs for the possible or actual addition of radio tuners, tape or wire recorders, and microphones.

7. Very careful assembly and matching of all units was followed by several days of testing and aural evaluation by several listeners including those purchasing the reproducer.

8. Careful instructions and hints were given both orally and in writing to prepare potential operators for the proper use and care of the instruments.

Side-by-side listening tests with other school and privately owned reproducers led to unanimous approval of the new machines by all those concerned, and the writer was invited by the music coordinator to give a lecture and demonstration using one of the new school machines at a convention of the Music Educators' National Conference in Salt Lake City, Utah in 1947. The reproducer used in the side-by-side comparison with the new Stockton machine was picked at random from the Salt Lake City schools. The choice was made at the last possible moment and the writer gave the demonstration without ever previously seeing or hearing the second reproducer. However, the lecture and demonstration was concluded successfully and some twenty-five out of approximately forty people present asked for plans and information to help guide them should they attempt such a construction project in their various school districts. Upon return to Stockton, the requested plans were completed, duplicated, and mailed to all those who had signed for them. Copies of these plans are reproduced in the appendix for whatever use they may be to the interested reader, and to illustrate for the purposes of this thesis the actual detail involved in constructing music reproducers.

It is doubtful if many of these plans were put into actual use because of the technical details involved in construction. The plans are practical and basically simple, but careful study of the plans by the reader will disclose the necessity of some previous knowledge of electronics, electricity, acoustics, diagram reading. We can assume that the music educators receiving these plans were adequately trained musically, but probably not at all experienced in most of these other areas.

Perhaps at this point the reader can see more clearly why the writer stated in the Foreward the basic concept that only those individuals who achieve a fusion of facts and ideas in the field of music, electronics, and acoustics can hope to achieve the adequate and artistic reproduction of music through electronic instruments. Commercially made reproducers are improving in design and quality every year. New types of disc and tape recorders are raising tonal standards to new heights. However, the individual who understands his equipment, whatever its quality, and operates it with insight as to its advantages and limitations will in that manner more adequately meet the aural demands of the critical listener.

It seems best to close this chapter with a brief evaluation of the results obtained with the various reproducers built for the Stockton schools. In the interest of brevity, the advantages and disadvantages will be listed under the general headings "negative" and "positive." Everyone can learn from negative as well as positive experiences, and most of these negative values were simply created by the environment and the always fallible human element.

Negative evaluations.

1. Teachers and students constantly dropped and

damaged pickup cartridges and stylus tips. (This was and is the most frequent and expensive repair.)

2. Busy teachers often left the machines turned on for unnecessary hours and at times even all night. Tube and condenser failures were frequent. Overall efficiency was reduced.

3. Since these machines were built just before the era of long-playing microgroove records, excessive surface noise on 78 RPM records caused most teachers to keep the treble control constantly turned low.

4. Volume expander circuits were misused as scratch suppressors by turning the expansion very high and the volume rather low. This reduced the soft and silent pasages on the records giving the impression of reducing scratch. The resulting overloading of the expander circuit added excessive dynamic distortion to the sound and brought consequent complaints from some listeners.

5. Parts purchased right after World War II were not of high quality and deteriorated rapidly with use.

6. The $135.00 cost limit was not high enough for really high quality reproducers. A more advantageous figure would have been $200.00 to $300.00.

7. Loudspeaker units were often placed behind desks or chairs cutting off the more directional high frequencies. Turntable units were sometimes placed on top of the speaker

cabinets as a convenient point of operation, causing vibration distortion.

8. In larger classrooms, teachers usually operated the machines standing to one side of or behind the speaker cabinet. Adjusting the tone and volume for their ears tuned to reflected sound often blasted the students with too much direct sound and caused some complaints, even from adjoining classrooms.

Positive evaluations.

1. Teachers and students became much more conscious of the need for better sound and they enjoyed listening to music with greater fidelity and dynamic contrast.

2. The reproducers exposed the poor quality of many records and performances helping the listeners to be more discriminating in their listening habits.

3. Fuller, richer, and louder sound levels held the attention of the class better than previous smaller machines with less loss of contact between student and teacher, when the latter alternated between lecture and record demonstration.

4. The machines outlasted the previous average life of older commercial models due to their more rugged construction and cabinet design.

5. The loudspeaker units have been incorporated into the construction of replacement models with more up-to-date

equipment at a cost saving and without sacrificing tone quality.

6. Dual and triple speed turntable motors and new pickups have been added without the necessity of changes in cabinet design, again decreasing the cost of modernization.

7. The idea of owning such reproducers for music classes seems to have become accepted as "standard practice" by many of the teachers. This is particularly true at the college level where students are more discriminating listeners.

8. Some of the advantages of better sound reproduction are now recognized all over the country through the growth of FM stations, the micro-groove record, and magnetic recorders. In Stockton schools we seem to have better reproducers and more of them than in almost any school this writer has visited in California, Nevada, Oregon, and Washington. It is hoped that we can maintain these standards as minimum in the music rooms of Stockton schools.

CHAPTER IV

AN ANALYSIS OF SOME OF THE REQUIREMENTS FOR THE ARTISTIC REPRODUCTION OF MUSIC

> Most of our knowledge is of the common-sense variety, gained in ordinary observation; very little is based upon experiment. Yet, where there is no experiment there can be no science. Furthermore, in a new applied science, such as the psychology of music, there is a vast amount of so-called "experimentation" that is neither scientific nor valid.
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> . . . .
>
> A survey of the experimental literature in psychology shows that experiments generally accepted as more or less scientific range from those which conform rigidly to these requirements to those which can scarcely be said to follow any of them. In this situation formative science can be tolerated on the ground that "doing the best we can" from time to time is often a stage preliminary to mastery. In all sciences we find such regions of exploratory effort.[1]

These statements by Mr. Seashore are meant to prepare the reader for what is perhaps the least scientific and most subjective area of musical understanding. If anything is to be accomplished, the reader and the writer need to keep an attitude that is alternately objective and subjective; objective toward new ideas and concepts in their initial form, and subjective in the evaluation of those ideas as one attempts to make use of whatever conclusions seem valid. Some conclusions may seem valid artistically and yet not be practical physically or psychologically. Practical

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[1] Seashore, *op. cit.*, pp. 44-46.

solutions to problems often are artistically disturbing to sensitive minds.

The following story illustrates the difficult, if not impossible, task of solving physical, psychological, and artistic problems with the same degree of success. Early in the 1930's Leopold Stokowski, as director of the Philadelphia Symphony Orchestra, recorded a long series of musical works for Victor records. New electrical recording processes had just been developed and these recordings were immediately popular as no other symphonic recordings were previously. Mr. Stokowski insisted on a radically different recording technique, a technique that was not new, however. In order for instruments to be heard properly on early orthophonic (pre-electronic) recordings, the instrumentalists gathered as close around the recording device as possible, and soloists performed closest of all in order to achieve some degree of harmonic and melodic balance on the recording. The use of sensitive electrical microphones made this arrangement unnecessary, but Mr. Stokowski insisted every solo instrument or section have its own microphone. As a result, every solo instrument or important section sounds "close" to the listener, is in the forefront of the sound, and is always dynamically and tonally superior to the other instruments. The average listener is greatly aided in his understanding of the music when his listening is guided through a musical score with such clear, broad, and definite lines of sound. The music takes on more presence and focus for that person and consequently greater meaning.

This procedure seems to be a practical solution to a psychological problem; that of getting the listener's attention and holding it. Artistically, the men in the orchestra were dissatisfied. They objected to a few instruments suddenly leaping out of the loudspeaker during a solo passage and just as suddenly being replaced by another instrument. They began to use as little dynamic contrast in their playing as they could, realizing that the recording technique would replace the dynamics through extra amplification and the constantly changing balance of the sound as it would be heard on the finished recording. In later recordings using the same techniques, the volume of the recorder was advanced slowly during certain solo passages, giving the listener the aural impression that the soloist began playing at a great distance from the listener and then rapidly came closer and closer as though floating in the air. While this aural effect might thrill a perceptive layman listener hearing only the total effect of the music, a sensitive symphonic instrumentalist, familiar with the more traditional sound of the passage could easily turn away from the recording in disgust! While Mr. Stokowski's experimental recording techniques may be accepted someday as standard practice, it seems safe to say that the majority of critical listeners would at present prefer more of a "concert hall" aural impression such as that described in the following statement:

> With this long playing Mercury Classics recording and others to follow with the renowned Chicago Symphony Orchestra under Rafael Kubelik's baton, Mercury has sought to mark a new departure in the art and science of orchestral recording--a departure marked by a striving for a truly realistic depiction of the sonority of the Chicago Symphony Orchestra performing in the magnificent acoustics of Orchestra Hall. Particular emphasis has been placed on achieving full dynamic range and on avoidance of tonal distortion resulting from attempts to create spurious effects of "ultra-wide" frequency range and brilliance. This recording and that of Bartok's Music for Stringed Instruments, Percussion and Celesta and of Bloch's Concerto Grosso (MG 50001) were made in Orchestra Hall, Chicago, on April 23-24, 1951. A single Telefunken microphone was hung 25 feet directly above the conductor's podium. The performances were recorded with Ampex tape recorders. The processing from tape to disc was done with Fairchild tape machines and amplifiers working through the Miller cutting head with the Fine-Miller Margin Control process of variable pitch and variable depth of cut--thus assuring a 100-percent reproduction in disc form of the recording originally captured on magnetic tape. David Hall was the recording director for these discs. C. R. Fine and George Piros were the engineers.
>
> FOR BEST RESULTS, this record should be played at FULL ROOM VOLUME. Only in this way will the quieter musical passages be heard in proper relation to the full orchestral climaxes. Owners of wide-range reproducing equipment are advised to set their bass and treble controls so that playback characteristic will be in accordance with the response curve published by the Audio Engineering Society.[2]

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[2] Mercury Classics, Program Notes, Moussorgsky, "Pictures at an Exhibition" ([n. p.]: Mercury Classics, 1951), p. 1.

The recording technique just described has resulted in one of the finest recordings this writer has heard up to this time.

Irving Kolodin, David Hall, and others have written books listing almost all available recordings up to the time of their publication and criticizing each in terms of artistic performance standards and how they are affected by the mechanical efficiency of the recording and the recording technique. The Kolodin book, called <u>A Guide to Recorded Music</u>, is strongly recommended to the reader as background reading for this chapter. He is a discerning musician, possessing a keen mind and occasionally an acid tongue. Through his frankness the reader benefits. Mr. Kolodin writes as follows:

> <u>Symphony in D Minor</u>. Beecham and the London Philharmonic Orchestra. C-Set 479--$5.50. (**fff$$). Stokowski and the Philadelphia Orchestra. V-Set 300--$6.50. (**ff$$). Mitropoulos and the Minneapolis Orchestra. C-Set 436--$5.50. (*ff$$$). Rhene-Baton and the Pasdeloup Orchestra. D-25685-9--$3.75. (*f$$).
>
> The weakness of the Mitropoulos performance, which only appeared a few months ago, were so pronounced that Columbia itself has superseded it with the splendid album of English origin. In a sense that misfortune of Mitropoulos (the faults of the set are essentially those of recording) was good fortune for the record-buyer, for one would not otherwise have had an opportunity to become acquainted with the fervent, controlled, magnificently sensible performance of Beecham. His splendid orchestra is heard at its best, and the recording is excellent. The Stokowski effort is eminently exciting, but equally exasperating, for the rush and turbulence of the music he creates is frequently agitated even more than Franck intended. For those who want to have the

> work in their collection for the convenience of friends, one may direct attention to the Rhene-Baton which includes all the notes at a minimum price. However economy is its only virtue, for the recording is dull, the surfaces noisy.[3]

It was stated earlier in this thesis that the full dynamic range of the orchestra or any other large performing group could not be recorded without reducing the volume of the loudest passages to avoid distortion in the recording and playback equipment. This statement is not literally true. The recorder volume control can be set to record the loudest sound to be produced by the performing group and then left alone. This may mean, however, that the softest passages will be too soft to be clear and audible above any background noises present in the recorder and/or the playback equipment. Ideally, the volume of the recorder should be monitored by someone familiar with the music and its dynamic contrasts.

The Alma Trio, a professional string trio consisting of pianist, violinist, and cellist, has occasionally performed at the College of the Pacific Conservatory in recital. On one of these occasions, the performers expressed rather keen dissatisfaction toward a group of recordings they had recently completed for a professional company. There was

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3 Irving Kolodin, _A Guide to Recorded Music_ (New York: Doubleday, Doran and Company, Inc., 1941), p. 148.

too much surface noise and the dynamics of the music had been rather seriously tampered with during the recording process, according to the Trio. The writer, during the course of the conversation, tried to point out the necessity of monitoring the recordings for mechanical and musical reasons, and then asked permission to record the evening's concert with the object of merely having a copy of the program for personal use.

When permission was granted, very careful preparations were made by placing the microphone above the players during their last rehearsal and checking a test recording carefully for the balance of sound between the instruments. The microphone was placed close enough to get a bright, full sound without destroying the blend of the ensemble. Musical scores of the music to be performed were obtained and studied closely for dynamic contrasts.

At the concert, the musical score was followed carefully. Very full crescendos were turned down very gradually for the duration of the crescendo. Sudden loud chords were compensated for by a quick but modest reduction of the volume just the instant before the chord was heard. All other adjustments were very gradual and unhurried to avoid any perceptable reflection of that adjustment in the recording sound. Where the volume was found to be too low or too high, the volume control was adjusted only when the musical phrase being recorded would seem to crescendo or diminuendo, and then that perceptable change was augmented slightly with the control to correct the recording level. Soon afterwards, the Trio expressed the desire to hear the recording played back on the writer's large test machine. The recording was done on a wire recorder so that there was almost no audible surface noise in the playback. The performers soon became so excited by the clarity and balance of sound harmonically and dynamically, that they stopped the playback and asked if their recent commercial recordings could be done over again on this recorder. They were particularly compliment-ary about the full dynamic range of the reproduced sound, not realizing that many changes had been made in the original dynamics. They were not aware of these changes because they were accomplished within the context of the music as it was performed. Observation has shown that there are relatively few symphonic radio broadcasts on the air where monitoring is accomplished smoothly and musically, and television sound monitoring is very inconsistent, at present.

The reader should now be aware that problems surround-ing artistic sound reproduction begin at the performance with such details as microphone placement and other attendant recording problems and techniques. The recording or broad-casting engineer who balances the sound heard through the microphones is potentially as important as the performers in the areas of tonal balance and dynamics. If the reader is to be the recording engineer as well as the playback operator, he needs to study microphones, acoustics, and recording devices very carefully. Recording music is a highly complex problem even with fine equipment.

Since most readers do not have access to performing groups for recording purposes, it is assumed that the major problems of artistic sound reproduction lie in the adequate playback of ready-made recordings from one source or another.

Some of the disadvantages of hearing reproduced music have already been mentioned in previous chapters. They are:

1. The element of seeing the performers is lacking, unless you are seeing television or sound movies.

2. The listener is aware that he is hearing a re-created performance, and except for television "live shows," the sense of participating with the performer is usually lacking.

3. The listening environment does not often fit the environment of the performance. Hearing the recording of a 300 voice chorus performing in a large, reverberant hall while looking at the confining walls of your living room is in a certain sense incongruous.

4. The reproduced music will only sound natural and realistic if it is played back at the same relative dynamic

level as the actual performance.

All of these points have been discussed previously except number three which seems obvious enough to accept without elaborate explanation. In light of these disadvantages, the first problem seems to be to prepare the listener psychologically for the listening experience.

Past experience at inviting people over to hear "some recordings" seldom resulted in stimulating musical listening. The social atmosphere of the occasion almost always interfered with the listening, and conversation would break out between records or even during extended soft passages. Inviting people to hear a certain musical work by name usually resulted in more attentive listening if the audience was initially favorable toward the invitation and that particular work. Mailing out a program to friends and acquaintances a week in advance seemed to give more of a program atmosphere to the evening and longer programs could be given without loss of attention.

It was this exact sequence of invitations that began the writer's series of weekly evening concerts that extended over a period of more than a year beginning in 1942. The audiences were mainly soldiers and their wives. This was especially true after the concerts were sponsored by the local U. S. O. This changing weekly audience plus a few persistent friends made up the test audience for the following experiments. This was certainly not a laboratory controlled study, as mentioned in the opening quotation of this chapter. In this situation, the writer found himself doing the best he could in an exploratory effort, trying to evaluate conclusions and in that manner set up new tests.

Experience seemed to indicate the advisability of speaking briefly to each audience to gather their attention before beginning the music each week. Dimming the lights in the room also seemed to help them relax and listen more attentively. Louder works seemed to be appreciated most late in the evening with a quiet work usually opening the program. The quiet work seemed necessary to set a sort of dynamic listening level in terms of the music volume in relation to the size of the room. Music that was too loud early in the evening usually offended the ears of some as being "overpowering." Yet on other occasions, the same composition late in the program could be reproduced at an even louder volume setting with no objections of any kind. Later on in the year, the programs were moved to one of the Army chapel buildings which seated approximately 250 men. In this environment, it was soon obvious that a larger room demanded more dynamic music to open the record programs effectively. Softer music was best placed in the middle for contrast.

Having begun with a discussion of preparing the listener, it seems expedient to discuss preparing the environment as well. Actually the two problems overlap in many ways. Soft lighting seemed to help relax the listener and made him more receptive; but some listeners still complained about loud passages occasionally being overpowering. Could the sense of being "overpowered" be partially because of the visual nearness of the sound source and the confining walls of the room? Suppose the lights were turned very dim so that just the barest outline of the room could be discerned, or even turned off so that the visual environment was completely removed? Would the music be more listenable, or less listenable?

At the next weekly concert, it was announced that the last number would be played in semi-darkness in order to permit the listener to transport himself mentally to whatever environment his imagination desired. A new recording was performed of "Harold in Italy" by Berlioz, recorded with considerable echo in Boston's Symphony Hall (Victor album #DM989). The recording was full and resonant and vibrant with vitality. The audience reactions after the concert were almost unanimously favorable. Many listeners, once the visual environment faded out of their conscious minds, literally felt as though they were sitting in Boston's Symphony Hall. The resonant acoustics of the hall seemed to fill the room and the room seemed to become the hall with the loudspeaker moved way back in one corner, where it was easily thought of as the orchestra. The impression was so realistic and yet inoffensive that for a time it became standard practice to play the last number in semi-darkness, always choosing recordings with considerable hall resonance. Studio recordings did not give the same effect. In fact, they gave somewhat the opposite effect if the recording was too lifeless acoustically.

Other methods of controlling the environment were to put the speaker unit and the phonograph operator out of sight and have ushers seat the audience and pass out programs. Bringing the lights down or up always signified the beginning or end of the program in order to avoid startling the audience with an unexpected burst of sound or leave them seated wondering if there were more program to come. The names of the orchestras and their conductors were left off the program to prevent even the suggestion of a famous orchestra and its home city from destroying the impression of the music being "here and now." These names were supplied unhesitatingly by the ushers, upon request.

All of the programs being discussed of necessity used 78 RPM recordings. Large radio transcriptions were not available and long playing micro-groove recordings were not yet on the market. Twelve-inch records played from four to five minutes on the average, which meant that as many as three or four record sides were needed to play one movement of a long symphony. The flow of the music and the listener's attention was always broken during record changes. Practice reduced the record changes to four or five seconds, but this still did not solve the problem. At this point a solution was found in standard radio station technique. Two identical turntables and pickups were purchased and mounted side-by-side with mercury power switches to avoid any mechanical noise interference. A third switch permitted either pickup to be heard separately or both pickups to be heard at once. With practice the operator could switch from the end of one record to the beginning of the next with almost no perceptible difference in the sound heard through the speakers. The second turntable was started with the pickup on the record a second or two before the first record had finished. Then the pickups were switched immediately at the end of the sound on the first record to keep the surface noise continuous and catch the first note on the second record. Since various records differed greatly as to the number of silent revolutions before the first groove modulations, each record change had to be practiced and memorized before each program.

The intensity of the surface noise sometimes varied radically from disc to disc and the volume of the tweeter speakers had to be adjusted instantly one way or another at record changes to keep the same amount of surface noise and help conceal the changes. Where the last record in an album needed to be turned over quickly, two albums were obtained in order to simplify the change and avoid a break in the sound.

Even with this degree of care and practice, changes were not always imperceptable, especially to critical listeners; but there was no question as to the desirability of the system. The reduction of mechanical noises and interruptions allowed the listeners to keep their thoughts on the music.

This system of two turntables does not always work to advantage in a living room for the reason that most listeners are distracted by the visual activities of the operator during record changes. Psychologically, most listeners do not like to see mechanical adjustments made during the course of the music. In the words of one listener, "It seems as though you are interfering with the performance and somehow altering it." This is a strange reaction when you consider how many mechanical and musical changes of one type or another have already affected the original sound even in the best reproduction. The listener should in reality be grateful for the human sensitivity of the operator producing the final playback. His musicianship and understanding of the playback equipment and the music can do much to enhance the beauty of the reproduced sound as he interprets it with the listener.

Mr. Stokowski writes of the desirability of this type of control in movie theaters of the future. He says:

> Techniques will be developed to control the maximum loudness of the music so as to make an appropriate level of loudness for each individual motion-picture house--because the acoustics in all theaters are different. The controlling of the maximum loudness will not be done in the projection room, as is usual at present, but at a point inside the theater which gives the average impression of the music as heard from every seat in the house. Because of the difference of acoustics in all theaters, it is doubtful if this control can ever be automatic. There is another reason why it should not be automatic. When a theater is only partly full, its acoustical conditions are different from those when it is completely full. All these differences should be adjusted, preferably by a good and experienced musician. The softest parts of the music must also be controlled, because if they are not soft and distant enough they will not be in the right relation to the rest of the sonorities, and so make the music sound dull and monotonous. On the other hand, if the soft parts of the music are too soft, the listeners will not be able to feel the emotional qualities of the music or its mystery--and these parts of the music will also be out of focus with the rest of the sonorities.[4]

Actually, the current three-dimensional picture and multiple sound track motion picture system called "cinerama" employs a sound engineer seated in the audience to control the multiple speakers and sound tracks. What better point of control and adjustment could there be than that which balances and proportions the final reproduction of the music just before it reaches the ear of the listener? This is

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[4] Stokowski, _op_. _cit_., pp. 256-7.

especially true if the recorded sound is known to contain mechanical or musical inadequacies.

Following one of the U. S. O. record concerts in the Army chapel building, a casual conversation revealed that all those still present felt a certain passage in the last composition had either been overly compressed dynamically during the recording process or the conductor had not realized fully the dynamic demands of the score. The suggestion was made that the operator, being out of sight, could subtlely amplify the passage to give it more dynamic range. The record was played again several times, each with varying amounts of added amplification. When the record was again played normally, everyone immediately agreed on the inadequacy of the dynamics. This led to experimental adjustments at subsequent record concerts. For example, extended soft passages were more adequately reproduced if the tweeter volume was reduced to lower the intensity of the needle scratch. No discernable difference in fidelity was noted due to the lack of intensity in the overtones of the soft music. The volume of soft passages was often reduced slightly to give greater dynamic contrast without the danger of over amplifying louder passages. Turning the volume of the tweeter above normal increased the brilliance of certain types of loud passages advantageously providing the clarity of the recording permitted such adjustment.

Noisy recordings sometimes required reduction of the tweeter volume during loud passages to reduce high frequency distortion. When any such adjustments were undertaken, they were always rehearsed ahead of time and, if necessary, control settings were written out on paper as a reference guide.

The reader should recall that volume expander circuits were in use from time to time which of themselves served the same general purposes, although with less musical discrimination due to the mechanical nature of the electronic circuit controlling the expansion.

A discerning reader may by this time have asked himself the following questions: (1) If reproducing music requires all these operating techniques, and I must be the operator, when do I have time to enjoy the music myself? (2) With new tape recordings and long-playing records, where there is almost no background noise and a greater dynamic range has been recorded, is it still necessary to monitor the playback?

The answer to the first question would depend somewhat on who asked it. The individual who attempts to study and understand mechanical and electrical equipment much as a fine instrumentalist studies his instrument will quite naturally use that instrument with ease and confidence. Certain individuals who "engineered" record concerts during the U. S. O. series never learned to understand the equipment. They seemed to have no feel for it and consequently often could not avoid obvious errors in operating technique. They were much like a poor actor who knows his lines but cannot deliver them effectively. Others seemingly never made mistakes of any consequence and their programs were always conducted smoothly and with dispatch. The operator who understands his equipment and knows the music will find certain adjustments suggest themselves as a result of that understanding and perception. The operator who does not understand his equipment, or perhaps the music, would best set the volume and tone controls at a median setting and let the machine do its job unaided or encumbered as the case may be.

The answer to the second question is already contained within the body of this thesis. No matter how perfectly music may be played and recorded, the playback will always be subject to infinite variations in atmosphere and environment. The acoustics of a room change in a marked manner when people enter or leave, when drapery is opened or closed, etc. The equipment itself is subject to variations in efficiency due to wear and tear, temperature changes, line voltage fluctuations, etc. The most important single factor in this answer lies in the varying "mental climate" of the listener. When the listener is passive and perhaps tired, music is often prefered at a low dynamic level without sharp contrasts. At a later time the same listener may play the same recording for a select and critical audience. The "mental climate" is now different and dynamic contrast, and crisp, clear sounds are the order of the day.

Musicians seldom perform a composition the same way twice. Many of their reactions during performance are psychological and emotional and therefore subject to variations which in turn affect the music. The reproduction of music can be more artistic if the listeners attempt to "participate" in the performance psychologically and emotionally, making as few or as many adjustments in the sound as seem necessary to meet the requirements of the music and the listeners at that particular time. To paraphrase a well-known slogan, it is possible to have, "Music as you want it, when you want it."

As a conclusion to this chapter, it is necessary to summarize the results obtained through the various procedures already described.

1. Because of wide variations in training and background, it is not always possible to satisfy the artistic and musical standards of the critical or the average listener.

2. While experimental recording techniques can produce unusual effects, the critical listener usually prefers a more traditional "concert hall" sound. This sound is best

obtained by a single microphone placed in front of, and above the performers.

3. Monitoring recordings is a job for a musician who understands the recording equipment and through it can proportionately balance the dynamics of the recorded sound with the aid of a musical score.

4. Those desiring to record music should study all types of microphones and recorders and gain as much experience as possible in acoustical application of microphone techniques.

5. Preparing the listener for the act of hearing recorded music is often as important as hearing the music itself. In general, the occasion should be treated as though a real performance were about to take place.

6. The environment of a small room usually requires medium loud to soft music for relaxed receptive listening at the beginning of a recorded program.

7. The environment of a larger hall suggests louder music to get the attention of the audience, then softer music as a contrast.

8. Removal of the visual environment for an audience in a receptive mood, by dimming or turning off all lights helps them to sense the acoustical environment of the original performance better and participate more completely in the atmosphere of the performance.

9. Conducting a recorded concert in the manner of an actual concert can often help provide a stimulating atmosphere for careful listening.

10. Careful mechanical preparations and planning by the operator should have as their goal the smooth presentation of the music as much as possible like it would be performed in a concert. Mechanical noises, unexpected gaps in the music, etc., must be avoided if the listener is to keep his attention on the music.

11. The careful monitoring of the playback can be as essential to the artistic reproduction of the music as it was at the original recording. The problems of the listener's environment and the dynamic and tonal balance of the reproduced sound can best be solved by the sensitive perception of a human operator.

12. Individuals who understand their equipment and know the music will find certain adjustments of the reproduced sound will suggest themselves as a result of that understanding. Individuals who do not thoroughly understand their equipment and/or the music would best set the controls at an average level and let the reproducer operate in that manner.

Experiments up to this time in the area of artistic sound reproduction are only mere beginnings in a challenging new field of human endeavor. Electronic music reproducers are present in almost every part of our daily living pattern. They are found in the home, automobile, office, factory, drive-ins, restaurants, service stations, waiting rooms, theaters, dance halls, skating rinks, auditoriums, etc.

The continued development and refinement of these instruments is inevitable, and the present interest in high-fidelity sound will continue to raise listener standards of perception and appreciation. Most of these developments have taken place since the advent of radio broadcasting in 1920. Consequently, there are many people who still think of music reproduction as something of a novelty without much to recommend it except convenience. It is interesting to note the continued existence of this attitude when experiments have indicated the most critical listeners hearing certain kinds of music while blindfolded cannot tell a live performance from a reproduced one if the finest equipment is used and listening conditions are carefully controlled. Mr. Bartholomew speaks of the problem as follows:

> It is no criticism at all to say that the sounds of high-quality systems for the electrical creation or reproduction of music are unmusical because they are "phonographic" or "mechanical." We know rather accurately beyond just what ranges of intensity and pitch the ear is incapable of hearing or discriminating, and apparatus can be built and has been built, although expensive, which will completely reproduce these ranges, so that the finest musicians' ears can be completely deceived as to whether the music is

"real," "canned," or "synthetic."[5]

Chapter V of this thesis attempts to show that certain future possibilities in sound reproduction are already possible, and that other possibilities await only the acceptance of tradition and practice.

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[5] Bartholomew, op. cit., p. 230.

CHAPTER V

THE POSSIBILITIES OF FUTURE DEVELOPMENT THROUGH TECHNICAL ADVANCES AND CONSEQUENT NEW AVENUES OF MUSICAL EXPRESSION

> There are those who fear the development of electrical instruments because they think they will make music mechanical. Exactly the opposite will take place. When the electrical instruments are relatively perfect, they will free musicians from our present constant preoccupation with the imperfections and technical difficulties of instruments. We shall be able to give all our feeling and thought to the inner essence of the music, because the instruments will respond with extreme sensitivity to every slight difference of feeling in the player and the music.[1]

Mr. Stokowski's statement should serve to help the reader look into the future with an open mind and a sense of anticipation.

The musician has almost always had to wait for the scientist to improve his instruments for him. The more complex instruments, become, the more the musician must depend on expert technicians to design and repair them. Musicians, frustrated by the inadequacies of certain instruments, have turned to the study of acoustics, metalurgy, wood and metal shaping, and lath work and attempted to improve and refine instrument construction procedures, develop new mouthpieces, key systems, etc. Their efforts have often been ignored by

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[1] Stokowski, *op. cit.*, p. 171.

those content to abide by tradition. Psychologists tell us that animals, human and otherwise, are born with an inherent suspicion of things that are new and unfamiliar. New ideas and beliefs in particular are accepted slowly and often reluctantly. Scientists have learned more in the fields of acoustics and electronics in the past twenty years than in all previous history. Is it any wonder that the acceptance and adoption of these discoveries is far behind the developments themselves?

The test reproducer described in detail in Chapter III produced what is termed monaural sound. That is, all the sound from all the instruments or voices heard in the original performance is condensed and reproduced through the relatively small single speaker system. When it is remembered that the performers are often seated in huge fan-shaped formations, it is easy to see why such a condensation of sound often "overpowers" the listener seated in the direct "beam" of the loudspeaker. The listener seated in the concert hall hears the performers through a greater spatial distribution of the sound sources. The violins in an orchestra will normally be heard from the left side of the stage seated in a section often twenty or more feet in length. The cellos usually will be heard from the right side of the stage seated in a like manner. If loudspeakers are mounted side-by-side completely across the end of the listening room, the sound will have more spatial distribution as described in Chapter III (page 70). However, since the sound was recorded through a single recorder, the violin and cello sections will be reproduced with equal volume by all loudspeakers and thus lose their directional qualities as heard in the traditional seating arrangements of the concert hall. Our ears sense the location of a sound source through the minute variations in sound intensity received by each ear and by the minute variations in the time it takes the sound to reach each ear. One ear is almost always closer to the sound than the other and therefore that particular ear would normally perceive the sound first and with the most intensity.

Following this basic explanation of binaural hearing to its logical conclusion, it becomes possible to visualize one microphone placed by the cello section and another by the violin section. Recording each microphone input separately with different recorders and playing the two recordings back with exact syncronization through separate loudspeakers which are spaced apart as were the microphones, will provide the listener with binaural sound reproduction. The music will have greater spatial distribution than monaural sound because the listener will perceive the directional characteristics of the sound. Dual channel tape recorders and disc recorders are now developed that accomplish this type of recording and playback very efficiently.

Increasing the number of microphones, recording channels, and playback units increases the spatial distribution and directional definition of the sound. The currently popular sound movie system called "Stereophonic Sound" uses such a multiple sound track system. At the present time, using a film strip is the most practical system for employing three or more sound tracks, due to the problems of syncronization.

Experimental broadcasts of binaural sound have already been completed by using two radio stations and two radio receivers to carry the separate sound channels.

Before World War II, the Walt Disney film studios in Hollywood, in collaboration with Leopold Stokowski, developed a revolutionary sound motion picture called "Fantasia." The sound reproducing equipment was designed with multiple channels and speaker systems for a binaural effect. During the last scene speakers placed around the walls of the theater were used to further increase spatial distribution of the sound in a system called "Phantasound." The sound system was effective but very expensive and far too advanced in design for most average listeners to appreciate. The idea was dropped commercially until the advent of television. The keen competition between television and the motion picture industries has forced movie producers to take advantage of binaural sound development and optical processes for adding depth to the image on the screen.

The most elaborate process yet developed in this area is that mentioned in Chapter IV called "Cinerama." A gigantic curved screen gives the illusion of depth to the picture, and multiple sound tracks and speaker systems, controlled by a sound engineer seated in the auditorium, provide binaural sound that syncronizes aural directional effects with the visual illusion of depth. The moving picture and television entertainment of the future will undoubtedly include such visual and aural illusions as standard procedure. "Cinerama" is at present simply designed to exploit its own powers of illusion. In time the medium could be developed and used to great effect in a more artistic manner.

New developments in electronic organs are making their tone and timbre more acceptable to discerning musicians. Touring groups such as the Bob Shaw Chorale are using these instruments because of their lack of bulk and weight. They never get out of tune and they can produce exceptionally low pedal tones without large pipes or bulky loudspeaker enclosures. This type of instrument can be used to reinforce small instrumental ensembles because of its flexibility and sustaining power, a distinct advantage that the more percussive piano does not possess.

It is a well-known fact that some of the low playing woodwind and stringed instruments in the orchestra are incapable of balancing the higher treble instruments in louder passages. The sound of an orchestra would be much fuller and richer if instruments such as the string bass could play louder. Experiments have already been made in this direction using various types of electronic amplification. When the proper solution is found, new techniques of orchestration can then be developed and the orchestra will become an even more versatile instrument. Mr. Stokowski tells us:

> Today we are on the verge of one of the greatest steps in the evolution of musical instruments that perhaps can ever take place--that is, the invention and development of musical instruments in which the tone is produced electrically, but is played and controlled through musicians' feeling, technical skill, and intuitive understanding. Some of these electrical instruments already exist, but are at present in a primitive stage.[2]

The addition of such instruments will open whole new areas of composition and performance. As Bartholomew says:

> Certainly these developments should be considered in a tolerant light. Sneers for the scientist have been popular among musicians ever since the beginning. Every advance in the mechanics of a musical instrument has probably been met by a storm of protest. But progress comes sooner through cooperation than otherwise. The question should not be, "Who dares thus to defile Art?", but, "What values do the various electrical methods for creating sound have for the future of music?"[3]

It is probable that with the increase of leisure time

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[2] *Ibid.*, pp. 169-70.

[3] Bartholomew, *op. cit.*, p. 230.

through labor saving devices, cultural activities will grow at an ever-increasing rate. Electronic musical instruments and sound reproducers will more and more provide an artistic outlet for the musically inclined and musically trained individual.

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APPENDIX

MUSIC REPRODUCER PLANS

PARTS LIST

Alternates are listed for each part in case the first listing is not available.

1. TURNTABLE--9" General Industries Model MX
   or--9" General Industries Model RX
   or--Green Flyer, Model D for 33-1/3 and 78 RPM

2. PICKUP--Astatic Model FP-18
   or--Astatic Model 508 with Nylon needle
   or--Astatic H P-16 or Nylon 400 for 16" transcription playback with 78 or 33-1/3 motor

3. AMPLIFIER--Bogen Model E-10 (for Phono. and/or P. A.)
   or--Bogen Model PV-10 (for Phono. only)
   or--Bogen Model PV-20 (at higher price, see unit 2-A)

4. LOUDSPEAKER--Jensen PM 12H - St - 476
   or--Cinaudagraph FB 1211
   or--Altec Model 600 (at higher price)

5. TONE EQUALIZER--(See equalizer diagram sheet)
   <u>Parts List:</u>
   1 small chassis--approximately 5" x 5" x 3"
   1 6C8G tube and shield, and grid cap
   1 octal tube socket
   3 face plates--Treble, Bass, Volume
   3 control knobs (match amplifier's)
   3 Potentiometers 2 - 500 M, 1 - 2 Meg.

RESISTORS		CONDENSERS	
No.	**Type**	**No.**	**Type**
1	5 Meg.	2	10 mfd cond.
1	500 M	1	100 mmf
2	250 M	2	.001
3	50 M	4	.1
1	2500 ohm	1	.05
1	1 M	1	.005

6. MISCELLANEOUS PARTS
   30 feet 2 wire rubber-covered cord
   6 feet 1 wire shielded pickup wire
   1 switch for the turntable motor
   1 pilot light, jewel type, to mount in motor box
   1 shielded phono. plug and receptacle
   2 speaker plugs to plug into the back of the amplifier
   1 piece loosely woven cloth to cover front of the speaker

About 3 feet of hookup wire, several parts mounting strips and the necessary tools. (See equalizer Unit 2-B)

TURNTABLE--UNIT I

Build the box out of 1/2" plywood

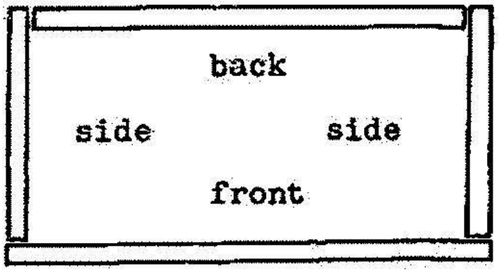
DIMENSIONS:

Top	18" x 15-1/4"
Sides (2)	12-1/2" x 7"
Front	18" x 7"
Back	17" x 7"
Motor Board	17" x 14"
Bottom (1/4" plywood)	17" x 14"

SUGGESTED FINISH:
Wood stain and varnish

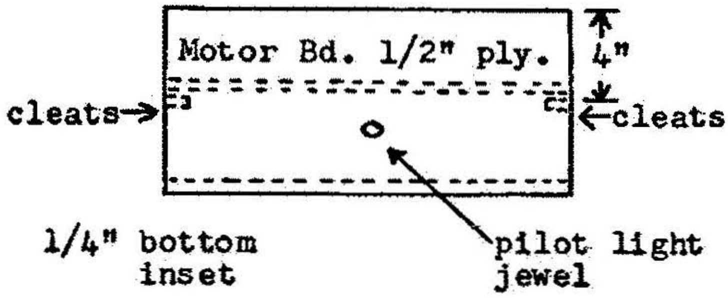
ASSEMBLY

Top View



FRONT VIEW

Front Panel



1. The top piece hinges from the top of the back piece to form the lid for the box.

2. The motor board rests on the cleats nailed 4" down inside the box.

3. Mount the turntable motor on the motorboard so that the turntable peg is equally spaced from the front to the rear of the box and from the left side.

4. Solder the 6 feet of shielded wire to the pickup. Replace the pickup wire completely if it is not shielded wire. (IMPORTANT: Excessive heating of the pickup cartridge with a soldering iron will permanently damage it.)

5. Mount the pickup on the right according to the directions given with it. Follow the directions carefully and to the letter, especially the height of the surface of the record from the motor board. (FP-18 is 1")

6. Mount the motor switch on the motor board to the right and toward the front.

7. Mount the jewel pilot light through the front of the box below the motor board. (See front panel diagram)

8. Wire the motor through the switch to about 6 feet of the 30 feet of cord listed on the parts sheet. Wire the pilot light directly to this cord, <u>not</u> through the motor switch.

9. The cord from the motor should plug into the back of the amplifier through a speaker plug, (see amplifier sheet Unit 2A) then the pilot light will burn continuously as long

as the amplifier is on and the turntable will operate only when the amplifier is on.

AMPLIFIER--UNIT 2-A

1. If you plan to use this machine for phonograph only, the PV-10 (see parts list) is recommended in conjunction with the equalizer, or at a higher cost and efficiency, the PV-20 by itself. If you will use the machine for public address with microphone and phonograph too, use the E-10 with the equalizer. (See equalizer sheet, Unit 2-B)

2. A radio can be played through any of the three amplifiers by running the output of the detector tube in the radio, or the voice coil leads of the radio speaker into the phonograph input in the amplifier.

3. If the output of the detector tube is used, use shielded wire to the amplifier and keep the lead as short as possible. This method while it is more trouble usually gives the best quality of sound.

4. If the E-10 is used and the microphone input on it is not needed, it is a good idea to ground the grid of the first tube and disconnect the wire from the center tap of the microphone volume control. If this is not done, care must be taken to see that this control is always in extreme off position when playing records.

5. If the equalizer is used with the PV-10 or the E-10 (See Unit 2-B) then disconnect the tone control in the E-10 or PV-10 by changing the wire from the center tap on the tone control to the unused tap on the control, or by removing the condenser connected to the center tap.

6. Wire the two speaker plug receptacles in the back of the amplifier so that the correct output goes to the speaker no matter which receptacle is used to plug in the speaker cord. Also, wire both so that either can be used to plug in the phonograph motor cord and get 110 v. A. C.

FOR EXAMPLE: Use pins 1 and 2 for the speaker output in both plugs, and pins 3 and 5 for the 110 v. A. C. to the phonograph motor. (This must, of course, go through the amplifier power switch). Then the speaker cord would be wired to plug in on pins 1 and 2 and the motor cord on pins 3 and 5.

7. Wiring the connecting cords between units in this manner makes it impossible to make a mistake in hooking up the machine for operation.

8. The pickup wire plugs into the equalizer or amplifier through a shielded pin jack.

TONE EQUALIZER--UNIT 2-B

1. IMPORTANT: Both the E-10 and PV-10 amplifiers have very inadequate tone controls. While this equalizer (see accompanying diagram) requires more technical knowledge to build and connect to the amplifier, the results obtained with it will more than compensate for any inconvenience in building it. The tone equalizer is as important in balancing the reproduced sound as the conductor was at the original performance.

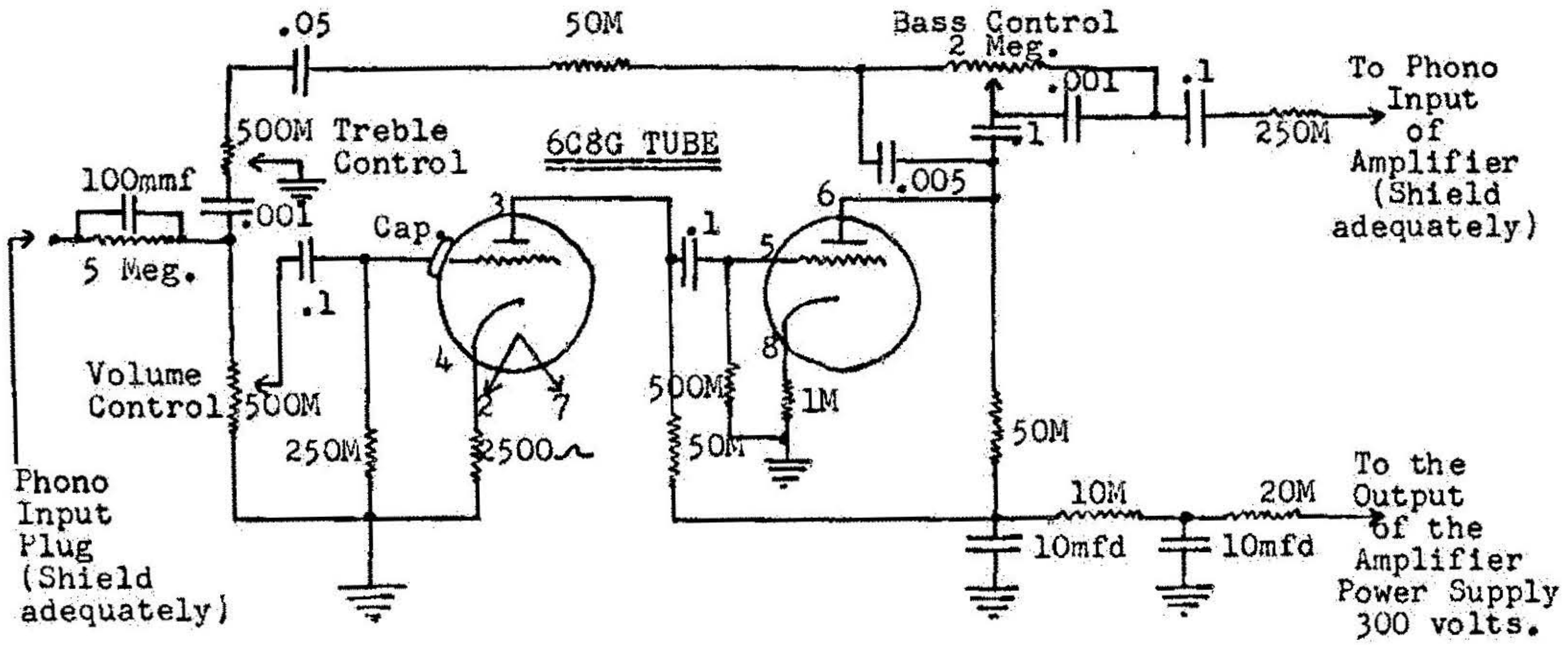
2. Build the equalizer in a small chassis that can be attached to one end of the main amplifier. Do not attempt to build it into the same chassis as the main amplifier. The hum picked up will cause no end of trouble.

3. Mount the three controls in order from left to right: volume, treble, bass, with the pickup input close to the volume control and the equalizer output closest to the phonograph input on the amplifier. Keep the lead from the equalizer to the amplifier as short as possible and completely shielded.

4. The power to light and operate the 6C8G tube in the equalizer is taken from the main amplifier as shown on the equalizer diagram.

5. Be sure to put a tube shield on the 6C8G.

TONE EQUALIZER CIRCUIT DIAGRAM



Pins 2 and 7 of the 6C8G should be wired to the filament power of the Amplifier 6.3v

SPEAKER BAFFLE BOX--UNIT 3-A

Use at least 3/4" 5 ply wood throughout!!

No.	Item	Dimensions
1	Front Panel	29-1/2" x 33"
1	Back Panel	27-3/4" x 31-1/2"
2	Side Panels	12-3/4" x 32-1/4"
1	Top Panel	12-3/4" x 29-1/2"
1	Bottom Panel	12-3/4" x 27-3/4"
1	Baffle Board	15-5/8" x 20-1/4"

SUGGESTED FINISH: Woodstain and varnish

1. Cut out the piece as shown in the front panel diagram 14-1/8" x 22-1/2".

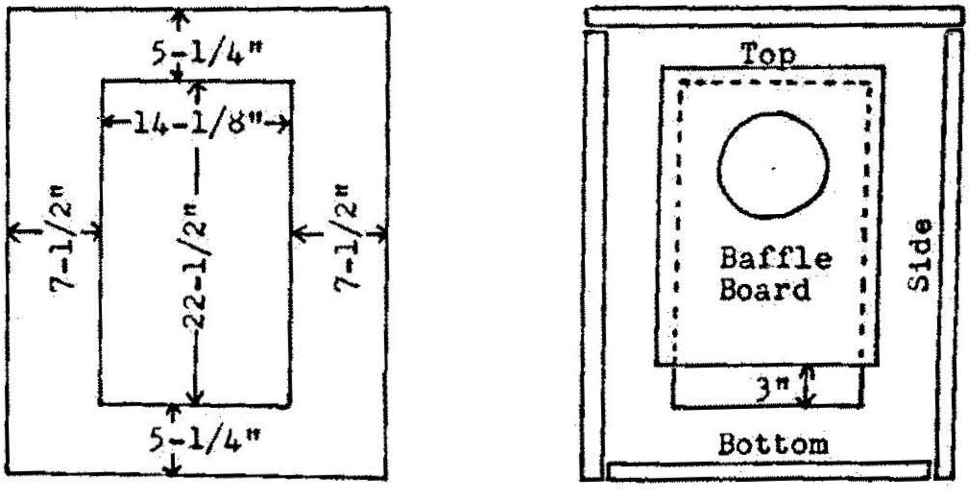
2. Cut a hole in the baffle board 10-3/4" in diameter as shown in the baffle board diagram 7-1/2" down to the center of the hole and 7-1/16" from each side.

3. Use 1-1/2" wood screws and glue in assembling the sides to the front panel. (See back view)

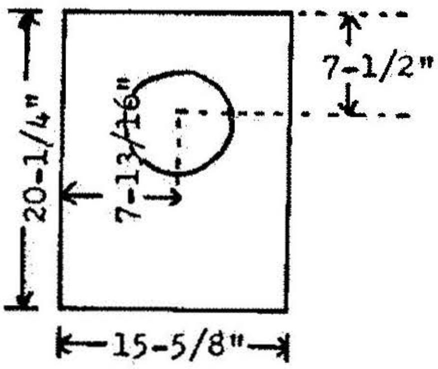
4. After assembling the pieces to the front panel, mount the baffle board against the inside of the front panel with the speaker hole up and with a 3" space between the bottom of the baffle board and the bottom of the hole in the front panel. (See back view)

FRONT PANEL

Front View — Back View



BAFFLE BOARD



5. See Unit 3-B for how to mount the speaker.

6. The back panel is put in last and cleats should be nailed 3/4" from the back of the box to hold the back in position until it is screwed in place.

7. Use plenty of screws and glue in assembly to make a strong, rigid box.

8. Tack several thicknesses of heavy cloth on the inside of the back panel in the form of a rectangle about 20" x 25" and one on the inside of each side panel about 6" x 25". An old blanket or rug furnishes good material.

9. If the machine is used in a room with a rug on the floor and curtains, etc., make the cloth pads a little smaller. A little experimentation will show how much you need. Too little makes the bass boomy, too much deadens the music.

10. A suggested method of covering the front of the speaker is to make a rectangular frame out of curved edge molding, a little larger than the hole in the front panel. Stretch and tack cloth over the frame and nail the frame to the front of the baffle.

LOUDSPEAKER--UNIT 3-B

FOLLOW IN THIS ORDER:

1. Assemble the baffle box according to the directions on the baffle dimension sheet.

2. Tack a piece of 1/2" mesh hardware cloth over the outside of the speaker hole in the baffle board.

3. Take 24 feet of the 30 feet of cord listed on the parts list and solder one end to the speaker as follows: Coded wire of the cord to the green voice coil lead, uncoded to the black. Wire the speaker plug to the other end of the wire so that the coded wire goes to the correct voice coil impedance. (The Jensen 12-H is 8 ohms.) See the diagrams sent with the amplifier for clarification. The other, or uncoded wire goes to ground or amplifier chassis through the plug.

4. If more than 24 feet is required between the amplifier and the speaker, a line to speaker transformer should be obtained. This transformer mounts on the speaker so you can run a 500 ohm line to the speaker and then step it down for the voice coil. (Follow the directions with the transformer.) The wire between the speaker and the amplifier can then be several hundred feet or more in length without loss of quality.

5. Mount the speaker inside the box over the round hole in the baffle board with 4 bolts. (Handle the speaker with great care; it is very easy to puncture the cone.)

6. Cut a notch in the bottom of the back panel for the speaker cord and tie a knot in the cord just inside the notch to protect the speaker from a pull on the cord.

7. Mount the speaker baffle on rubber cushions or casters to help eliminate floor vibration.

8. A handle can be mounted on each side of the baffle box to aid portability.