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A COMPARISON OF THE INTELLIGIBILITY OF COMPRESSED, CLIPPED,

AND NON-LIMITED QUIET AND NOISY SPEECH SIGNALS

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CHAPTER I

INTRODUCTION

In electronic technology the term "limiting" refers to the intentional restriction of the maximum amplitude that a signal is permitted to attain. Signal limiting became mandatory in the broadcasting industry when the Federal Government introduced legislation which closely regulated the maximum signal power that a station was permitted to radiate.

Abrupt peak clipping was widely adopted as a simple and effective method of restricting signal power, but limiting by this method resulted in the generation of undesirable harmonic distortion. Consequently, in the early 1930's the broadcasting, recording, and other communication industries began to turn instead to the use of automatic gain control amplifiers for limiting purposes.

An automatic-gain-control (agc) amplifier is defined as "any amplifier which, without human intervention, acts to change the amplification in a patterned manner" (30). There are two general classes of automatic-gain-control amplifiers. The distinction depends upon the effect of an amplifier on the dynamic range of a signal. An agc amplifier that extends the dynamic range of a signal is referred to as a gain-increasing amplifier or expander. An agc amplifier that decreases the dynamic range of a signal is referred to as a gain-reducing amplifier or compressor (30).

Grimwood (22) further subdivides gain-reducing amplifiers into compressors and limiters, which are distinguished principally by the numerical values of their compression ratios. Compressors are characterized by fairly low compression ratios ranging from two-to-one to, perhaps, five-to-one. In contrast, limiters are characterized by substantially higher compression ratios such as ten-to-one or more. Both compressors and limiters perform as conventional amplifiers for low-level signals but offer reduced gain to signals exceeding an arbitrarily selected level. This level is referred to as the threshold of compression.

There are three important effects of signal compression. First, the absolute magnitude attainable by large-amplitude signals is restricted or limited. Second, the dynamic range of the signal is reduced. Third, small signals below the threshold of compression are amplified more than larger signals which exceed the threshold. These three effects have been thoroughly exploited by the broadcasting, recording, communication, entertainment, and sound reinforcement industries and, to a more limited extent, by the hearing aid industry.

Compression has become the preferred method of restricting maximum transmitted power in commercial braodcasting. Compression, in contrast to peak clipping with its associated distortion, makes possible a precisely limited output level while maintaining the signal waveform. In amateur and commercial systems compression is often used not only to limit output but to improve the signal-to-noise ratio by boosting the level of the weak sounds above the level of the noise existing in the communication channel.

The use of compression amplification is particularly advantageous in the sound recording industry since it enables the recording engineers to accommodate the wide dynamic range of the program material to the inherently narrower dynamic range of the recording medium without introducing serious distortion. The recently introduced Dolby System (16), in which compressors and expanders are combined to effect a substantial improvement in signal-to-noise ratio, has virtually revolutionized the industry.

Compression amplification is also used to govern the range of sound intensities reproduced in a theatre, arena, or transportation terminal by limiting potentially intolerably loud sounds while at the same time amplifying weaker sounds which would otherwise be masked by high ambient noise levels. The net effect is an improvement in signal-to-noise ratio.

The hearing aid industry has also recognized certain advantages of compression amplification. For more than thirty years compression circuitry has been available in hearing aids as a means of limiting the acoustic power reaching the ear without incurring the distortion associated with peak clipping. Surprisingly, however, the majority of hearing aids still employ clipping as their method of restricting maximum output power.

It remains to be demonstrated whether or not the improvement in signal-to-noise ratio made possible by compression amplification in other applications can be realized with hearing aids. The uncertainty of the matter stems from consideration of the circumstances under which aids are ordinarily used. These conditions differ substantially from those encountered in the applications mentioned previously.

Specifically, in virtually all of the above applications, the noise exists either in the transmission channel or recording medium, or at the signal's destination. In recording, for example, the noise arises from the granular composition of the film emulsion, the surface noise of the disk, or the hiss of the tape. In a bus terminal, the noise is attributable to the hubtub of the crowd and the arrival and departure of the vehicles. In these and all other instances in which signal compression has proved to be effective in improving the signal-to-noise ratio, it has been possible to modify signal level relationships prior to the introduction of the noise. In the situations in which hearing aids are generally used, however, the noise exists in the environment and occurs coincidentally with the signal so that both the signal and the noise undergo compression.

The broad, unsubstantiated, and unqualified claim of "improved hearing in noise" for users of hearing aids employing compressor circuitry has been circulated within the past ten years by several manufacturers of these instruments. Consideration of this claim brings to mind several questions. First, is the alleged improvement relative to the results obtained with peak clipping, or is it relative to non-limited speech? Second, is this improvement observed for both favorable and unfavorable signal-to-noise ratios? Third, can it be assumed that "improved hearing" refers to the intelligibility of speech, and fourth, is this improvement the result of a superior signal-to-noise ratio offered by the compressor?

The available information concerning the effects of limiting on the intelligibility of speech when noise enters the system falls into two categories. The first category encompasses a few studies in which the effect of peak clipping on the intelligibility of speech was investigated in military communication systems. It was found that while peak clipping in quiet improves speech intelligibility by allowing the average level of the transmitted signal to be increased, clipping in noise is deleterious to intelligibility. No information is available as to the effects of compression amplification on the intelligibility of speech when noise enters the amplifying system.

The second category consists of a few scattered opinions confined to speculation about the effect of compression and peak clipping on signal-to-noise ratio. Silverman, Taylor, and Davis (55) seemed to suggest that compression would result in a better signal-to-noise ratio than that afforded by peak clipping. Kretsinger and Young (27) predicted that "a unique masking problem" would occur if noise was allowed to enter the compressor along with the signal. Rutherford (51) held that the speech-to-noise ratio at the output of a compressor would remain the same as that existing in the environment, but that the signal-to-noise ratio at the output of a clipper would be degraded. Krebs (26) also suggested that compression would have no effect on the signal-to-noise relationship. He went on to comment, however, that if the background noise was softer than the speech signal, compression offered an advantage relative to peak clipping, but if the background noise was the louder signal he predicted that clipping would be more advantageous than compression. Ling (36) stated that in order to overcome the problem created by background noise for children wearing hearing aids he excluded those aids employing compression circuitry and used high acoustic input with low-gain instruments. He claimed that these particular circumstances improved the child's ability to differentiate between background noise and speech.

In view of the conflict of opinion, the lack of available evidence in the matter, and the importance of the implications that the aforementioned considerations hold for hearing aid use, an experiment was conceived for the purpose of exploring the comparative intelligibility of compressed, clipped, and non-limited speech in quiet and in noise. The experiment called for the use of three different lists of equivalent CNC speech materials presented under these experimental conditions to eighteen normal-hearing subjects at -5, 0, and +5dB signal-to-noise ratios and in quiet.

The following chapters are devoted to a review of the literature concerning compression amplification and peak clipping, a description of the experimental apparatus and procedures used in the conduct of the experiment, and a presentation and discussion of the results.

CHAPTER II

REVIEW OF THE LITERATURE

Introduction

Every system involved in the transmission, recording, or reproduction of sound must confine its operational parameters within well defined limits. These limits may be determined by the saturation level of the circuit elements within the system, by federal regulation, or by the tolerance of the human ear. The intentional restriction of the maximum amplitude that a signal is permitted to attain is referred to as signal "limiting".

Methods of Signal Limiting

There are presently two commonly used methods of signal limiting. These are peak clipping and compression. The methods are distinguished by their effect on the amplitude of a signal. As the term implies, a peak clipper reproduces the waveform of a signal up to a preset level, but any portion of the signal that exceeds this level is not reproduced or is "clipped" (12). The resultant waveform may appear as a flat-topped wave or, upon extensive clipping, as a square wave. Either the positive or negative peaks of the signal, or both peaks, may be affected. The most common peak clipper used in speech transmission systems offers symmetrical clipping in which the amplitude of both the positive and negative peaks of the signal is limited. In contrast, a compressor simply reduces the amplitude of any signal which exceeds a preset level without changing the waveform.

The Clipper

The most common means of achieving peak clipping is by relying upon the ceiling or saturation point inherent in any amplifier. This level is sometimes referred to as the threshold of clipping. Beyond this level sufficient power cannot be supplied to the circuit elements to enable them to reproduce the peaks of the signal.

Below its threshold of clipping a peak clipping performs as a conventional amplifier exhibiting linear gain characteristics. A signal which exceeds the threshold of clipping overdrives the final output stage of the amplifier and is limited. This inability of a clipper to reproduce the waveform of an input signal faithfully at its output results in distortion. If a sinusoidal input signal is limited by a clipper the output waveform is no longer sinusoidal but is composed of numerous frequencies. The additional frequencies result in a broadening of the frequency spectrum and the production of noise. In order to reduce the breadth of the frequency spectrum and to eliminate a portion of the noise, a low-pass filter is commonly used in conjunction with a peak clipper. Despite such attempts to reduce the distortion products, peak clipping often degrades the intelligibility of speech.

In summary, the simplicity of the circuitry involved in peak clipping has made it a relatively inexpensive and practical method of signal limiting. The distortion associated with clipping, however, has restricted its use and in many cases has led to its replacement.

The Compressor

A compressor is more expensive than a clipper because it requires complicated circuitry. This circuitry also takes up additional space and this may become an important consideration in hearing aid design. The conventional compressor performs its task by means of a feedback circuit (10, 60). The input signal is applied to a variable-gain amplifier which in turn drives a fixed-gain amplifier. The output of the fixed-gain amplifier is sampled and fed back to the variable-gain amplifier reducing its gain. Very recently, innovations in circuitry and components have permitted the construction of compressors of vastly improved performance characteristics involving a different principle of operation, but these are not yet in common use.

There are two types of compression circuits. These are the compressor and the limiter. Grimwood (22) differentiates between the two instruments by their compression ratios. Compressors characteristically employ relatively low compression ratios of 2:1 to, perhaps, 5:1. Limiters exhibit compression ratios of 10:1 or more. Both compressors and limiters share basic characteristics which are important to their functioning. These have to do with certain aspects of their input-output functions and with the values of their time constants. These topics are treated below.

Input-output functions. Figure 1 shows the input-output functions of several typical amplifiers. Line A-B-F represents the input-output function of a conventional linear amplifier. Points A-B-C define the input-output function of one kind of compressor, points A-B-D represent the input-output function of a limiter, and line A-E defines the function of another variety of compressor. The three important charact-

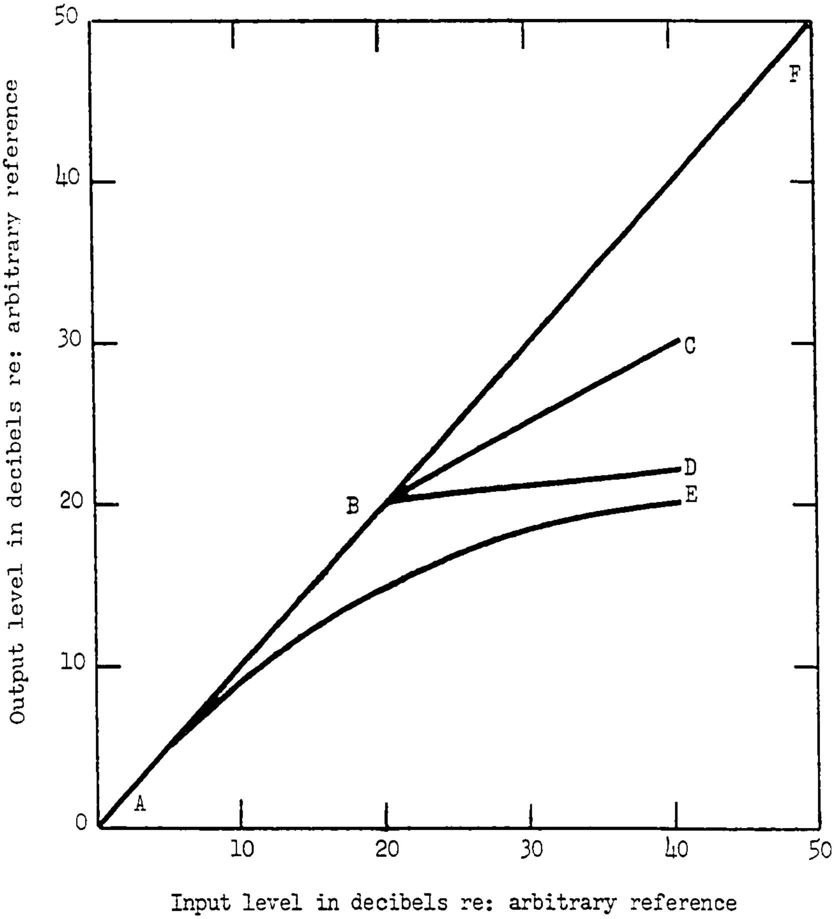


Figure 1--Input-output functions of a typical conventional amplifier, constant ratio compressor, variable ratio compressor, and limiter.

A-B-C = Constant ratio compressor

A-B-D = Limiter

A-E = Variable ratio compressor

A-B-F = Conventional amplifier

eristics of such functions are the threshold of compression, the range of the compression region, and the slope of the compression region or the compression ratio.

The threshold of compression. Grimwood (22) defines the threshold of compression as that point at which the "output level is one-half decibel below the uncompressed output level." In Figure 1, point B represents the threshold of both the compressor and the limiter. For low-level signals, either the compressor or the limiter performs as a conventional amplifier exhibiting linear gain characteristics. Once the output level of the unit exceeds the threshold of compression, however, the compression circuit is activated and the output signal is reduced or compressed.

Range of the compression region. Grimwood (22) defines the range of the compression region as the "useful working range" of the compression circuit. The lower end of the range is defined by the threshold of compression and the upper end of the range is determined by the saturation level of the instrument. Grimwood states that this region can be expressed as an input range or as an output range. In Figure 1, line B-C represents the useful working range of the compressor and line B-D represents the working range of the limiter. For both instruments the input range is 20dB, but the output range for the compressor is 10dB and that for the limiter is 2dB.

Slope of the compression region or the compression ratio. Grimwood (22) defines the slope of the compression region as the output range divided by the input range. For the compressor in our example the slope of the compression region is 10dB divided by 20dB or .5. For the limiter the slope of the compression region is 2dB divided by 20dB or .1. The slope of the compression region can also be expressed as a ratio of the range of the input signal (beyond the threshold of compression) to the range of the output signal. For the compressor, the compression ratio is 20dB-to-10dB or 2:1, and for the limiter it is 20dB-to-2dB or 10:1.

A compressor may employ either a constant or a variable compression ratio. In Figure 1, points A-B-C define the input-output function of a constant ratio compressor circuit. In this example, once the threshold is exceeded, a fixed ratio is maintained between the input and output signal levels regardless of the level of the signal. Curve A-E represents the input-output function of a variable-ratio or curvilinear system. In this example, the compression ratio increases progressively as the intensity of the input signal is increased. In other words, as the input signal level increases in intensity, progressively more limitation is imposed upon the amplifying system. The curvilinear input-output function is typical of that employed in currently available hearing aids using automatic gain control circuits.

Time constants. Unlike the peak clipper which acts instantaneously upon the application of a signal peak, the compressor is designed to decrease its gain rapidly upon the sudden application of a signal and to increase its gain slowly upon the cessation of the signal (22). The time required for the amplifier to complete a specified change in gain from an uncompressed value to a compressed steady-state condition upon the application of a signal is referred to as the attack time or operating time(22). The time required for the same relative change in gain to occur from the compressed value to a steady-state uncompressed condition upon the cessation of the signal is referred to as the release time (22). The attack and release times are routinely specified as the times required for the compressor to achieve 63 per cent of the final amplitude of the steady-state signal level.

Because a typical compressor utilizes a feedback circuit with its inherent time constants, a delay in the gain-changing process occurs with respect to the onset of a signal peak. If the duration of the signal peak is short compared to the operating or attack time of the circuitry, the signal will occur before gain reduction can be accomplished and will escape compression (6). The selection of time constants is, therefore, an important consideration with many kinds of signals. For material which fluctuates rapidly in level, an extremely fast attack time is required. The shorter the attack time, the fewer the number of cycles of a signal which are reproduced with amplitudes greater than the desired output of the amplifier (6).

Rapid changes in gain which occur when a compressor processes a fluctuating signal can, themselves, create a form of distortion. With a fast attack time, however, the distortion lasts such a short time that it is not disturbing and is often imperceptible (56). Similar distortion associated with the release time causes a low-frequency thump which is usually eliminated by filtering (22, 42, 56). If the release time is too long, another problem is created because the time that the amplifier remains in the compressed state may exceed the duration of the intense portion of the signal. This results in a prolongation of reduced gain during the reproduction of weaker sounds which should have been amplified in order to maintain audibility. These inaudible passages are referred to as program gaps (22). Reduction of the release time for the purpose of eliminating program gaps, however, may result in an audible restoration of gain during quiet periods. This effect is referred to as pumping (22, 41).

Uses of Peak Clipping and Signal Compression

Both peak clipping and compression have been utilized in the broadcasting, recording, communication and sound reinforcement industries and to an extent, in the hearing aid industry. The practical applications of peak clipping and compression amplification in these areas will be discussed in the following sub-sections.

Broadcasting Industry

Commercial broadcasters must restrict the radiated power of their transmissions according to well defined limits (6, 24, 31, 63). Within these limits, it is desirable to maintain the highest possible average signal level to avoid masking by noise which is inherent in the transmission system, without overmodulating the carrier signal. For example, when speech and music are included in the program material, the peak voltages may be 8dB or more greater than the average level of the signal (6). If these peaks are kept below the 100 per cent point of modulation, then the average level of the signal is reduced accordingly. Although use of this level would prevent overmodulation, it would, instead, create a masking problem, since the weaker portions of the signal would be lost in the noise of the transmission system.

If the average level was boosted, however, and the peaks of the signal were allowed to overmodulate the system the importance of the result would depend upon the frequency of occurrence of the peaks. Infrequent overmodulation would cause minimal harmonic distortion and would therefore be tolerated. More frequent overmodulation, however, would cause noticeable distortion and could possibly cause a breakdown in transmission because the signal voltages might be great enough to destroy circuit elements within the transmitter (6). Constant over-modulation could lead to the most serious problem which is now regulated by the Federal Communications Commission, that of interference with adjacent transmission channels (10).

In the early years of broadcasting, the level of the transmitted signal was controlled manually by a program operator. Because of the rapid transitions in the levels of speech and music, the manual method of volume control was inadequate (56). The earliest practical automatic limiter was the peak clipper. Peak clipping effectively limited those portions of a signal that exceeded the saturation point of the clipper, allowing the average level of the signal to be increased without increasing the possibility of overmodulation by the signal peaks (6).

Peak clipping proved to be effective when the signal peaks were infrequent, and the occasional harmonic distortion introduced by the clipper was tolerable. When the peak clipper was utilized on program material which contained repeated signal peaks, however, the attendant distortion caused an unnaturalness of the reproduced signal and, often, a signal which was unintelligible.

The inadequacy of the program operator to achieve effective manual control of program level, and the distortion inherent in limiting by peak clipping, set the stage for the development of a more satisfactory method of automatic volume control (6, 56, 63). The compressor made it possible to maintain the highest possible average transmitted signal level without incurring the distortion associated with peak clipping. The compressor automatically reduced its gain upon the application of what otherwise would have been excessive signal peaks. The resulting waveform was not distorted but was merely a reduced version of the original signal, and, immediately after a signal peak, full gain was restored so that the weaker portions of the transmitted signal were boosted above the noise inherent in the channel (11, 22). A programmer with a foreknowledge of his program material and an awareness of the flexibility of compression amplification could utilize compression as an effective and distortion-free method of maintaining the highest possible average level of the transmitted signal.

Recording Industry

The wide dynamic range of music and speech have also posed a major problem for the recording industry (12, 22). As Grimwood (22) remarked, "the range of sound levels to which the ear is sensitive is much greater than the range which can be linearly accommodated by any method of sound recording." The upper limit of recorded sound is dependent on the maximum permissable level of modulation. The lower limit is dependent on the noise characteristics of the recording medium (22). In sound-on-film, tape or disk recording, weaker sounds, if insufficiently boosted, are masked by the noise inherent in the recording channel or medium.

The earliest attempt at solving this problem involved manual adjustment of the signal. A highly trained program operator who was thoroughly familiar with the program material "rode gain", reducing the higher levels and boosting the softer passages in an attempt to accommodate the program material to the dynamic range of the medium. But the constant fluctuations of the signal often made manual control ineffectual. French and Steinberg (20) reported that as many as ten speech sounds may occur per second, and Fletcher (19) observed that these sounds may fluctuate over as much as a forty-decibel range for one speaker and fifty decibels between speakers. Peak clipping assisted the operator by limiting excessive peaks, but the resulting distortion created an unnaturalness in the quality of the recorded materials.

The use of an amplifier which was capable of automatically changing its gain so that its output level was limited below the point at which overmodulation occurred allowed the operator to utilize a higher average level of recording without the danger of distortion caused by clipping. Furthermore, the compression of the dynamic range of the input signal into a narrower range at the output of the amplifier facilitated the recording of a wider dynamic range of sound levels (22).

Grimwood (22) reported that compression of the dynamic range of the material to be recorded also restored naturalness to the recording. This author claimed that room acoustics and the monaural characteristics of the recording resulted in a volume range greater than that of the original sound. The compressor reportedly restored the volume range to a more natural state.

Recently, Dolby (16) introduced a system of recording which results in a superior signal-to-noise ratio, a wider dynamic range and reduced distortion. Dolby utilized a compressor-expander pair operating in each of four bands of frequencies. As passages of soft music passed through the compression circuit the weaker portions of the signal were boosted in order to prevent masking of the low-level sounds by tape noise. Upon playback the expander restored the dynamic range of the signal to its natural state. Thus the system improved the signal-to-noise ratio while preserving the dynamic range of the signal.

The Dolby System is now used almost universally by professional recording companies. At least one manufacturer of high-fidelity equipment for home use has incorporated a simplified Dolby System in a tape recorder-reproducer.

Motion Picture Industry

In the motion picture industry compression amplification has also been used in recording film tracks. Signal compression prevented overmodulation by signal peaks while at the same time it boosted the weaker portions of the signal above the film noise (50).

In the reproduction of a film track in a noisy environment, particularly that of a theater where audience noise was high, the development of a method of boosting the signal level so that all of the program material could be heard above the audience noise was essential. It was impossible to increase the volume of the sound track in its entirety because this resulted in signal peaks which were intolerable to the ear of the listener. Peak clipping limited the signal peaks and allowed the average level of the signal to be boosted, but, at the same time, it introduced distortion which was noticeable to the audience.

Again, the use of compression amplification solved this problem without the distortion associated with peak clipping. It prevented loud sounds from becoming intolerable, while at the same time it raised the weaker sounds over the noise of the audience.

Other Communication Systems

During the last two decades compression has also replaced peak clipping in apparatus such as public address systems, intercom systems, and sound reinforcement systems. In these devices compression circuitry is used to maintain a relatively uniform level of output regardless of the fluctuations of the input signal caused by talker variations, distance from the microphone, changes in the material presented, and other sources of signal variation. The compressor further serves to maintain a favorable relationship between the output level of the system and the ambient noise level of the environment into which the signal is introduced. Compression automatically provides protection against sudden loud sounds by reducing the amplitude of the signal while at the same time it achieves audibility for the faintest sounds by affording them full amplification. In amateur and commercial communication systems compression is often used not only to limit output but to improve the signal-to-noise ratio by boosting the weak sounds above the level of the noise in the communication channel.

Limiting in Hearing Aids

In 1947, Davis (13) defined a hearing aid as "any instrument that brings sound more loudly to the listener's ear. It may simply collect more sound energy from the air; or it may prevent the scattering of sound during transmission; or it may provide additional energy usually from the batteries of an electrical amplifier." In a discussion of the objectives of a hearing aid, Davis (13) states that, "...a hearing aid should deliver sounds loudly enough to be heard easily, but without discomfort.... Distortion of the original pattern of sound should be introduced only to the extent that it assists in bringing to the listener speech that is at the same time intelligible, comfortable, and of pleasing quality."

Aspinall (4), in discussing the design of commercial hearing aids, suggested that the two basic qualities of a hearing aid are that the intensity or loudness of the signal is of sufficient magnitude to deliver the sound to the listener's ear and that the quality of the reproduced signal is good. With the strides made in electronic and electro-acoustic engineering in the 1940's, the problem of insufficient loudness of an amplified signal was overcome. The conventional hearing aid could produce as much sound as the human ear could tolerate. It was immediately recognized that instruments which were capable of reproducing sound at extremely high levels must have a volume control. Mandl (40) stressed that a hearing aid user must be able to regulate the loudness of sounds coming through his aid in order to meet the ever changing conditions of his environment. He further stated that the maximum volume must be limited to protect the ear from intolerably loud sounds.

Even early hearing aids offered inherent protection against intolerably loud sounds, since the amplifying circuit could deliver only a limited amount of current through its final stage of amplification (3, 13). This common method of limiting acoustic power was named according to its effect on the signal waveform. Any portion of a signal which exceeded the ceiling or saturation point of the output stage of the amplifier was not transmitted. The resulting signal appeared to be flat-topped or, upon severe over-driving, it resembled a square wave. The process was, appropriately enough, called peak clipping. A certain amount of clipping was tolerable, although it obviously distorted the signal waveform.

Davis (13) suggested that for those individuals with a severe hearing loss, a considerable amount of simple peak clipping did not reduce the intelligibility of speech even though it made the voice sound harsh, rough, and unnatural. Extreme clipping did, however, affect the intelligibility of the signal.

In 1936, the Multitone Company of England incorporated compression circuitry into a desk-type amplifier called the Reactor (46). The inclusion of this feature in a wearable hearing aid was not possible at that time because the space available in a wearable device was not adequate for the incorporation of the additional bulky components necessary to effect compression (47).

As early as 1943 researchers were becoming aware of the problem of designing hearing aids for the hard-of-hearing individual with a sensori-neural hearing loss (37, 45, 47, 48). Littler (37), in discussing the types of hearing losses which demanded specific requirements for hearing aids, stated:

> It has long been realized that patients suffering from appreciable inner-ear deafness, signified by appreciable bone conduction hearing loss, hear sounds which are only 20 to 30dB above their minimal audible threshold as very loud sounds, and it has been believed that such patients, while having subnormal hearing for weak sounds, have almost normal hearing for loud sounds .... When an inner-ear patient uses a hearing aid whose amplification is sufficient to make the weaker sounds of speech audible, the louder sounds become disagreeably loud.

Littler concluded, "In order to lessen the amplification of loud sounds as compared with the feebler sounds, it is necessary to incorporate some kind of automatic volume control."

Pothoven (49) also advocated automatic volume control. He stated that it is "...technically more difficult to provide a wearable hearing aid with compression amplification (extra tube and condensor), but it gives better results than peak clipping."

In 1948, the Multitone Company of England introduced the first commercially available wearable hearing aid to employ compression circuitry. The "Monostat" as it was called, was followed shortly by a second wearable AVC hearing aid, the "Selector" (46).

Although in 1949, compression circuitry had not been employed in any domestic wearable hearing aids, experimentation was being carried out with group hearing aids that used this principle. Harrison (54), at the Central Institute for the Deaf, constructed two group hearing aids at the request of the Subcommittee on Group Hearing Aids of the Committee on Deafness of the National Research Council. The advantages claimed for these aids were that the AVC circuit assured a teacher that her voice was not too loud for the students and that the students' fear of sudden, intolerably loud sounds was eliminated. These advantages were considered to offer definite psychological benefits to both the teacher and her students.

In the early 1950's, some of the problems in the design of wearable hearing aids with compression circuitry were overcome and domestic manufacturers of hearing aids began to incorporate compression circuitry into their lines of aids. Initially, the literature describing these instruments suggested simply that the use of automatic volume control circuitry eliminated constant manipulation of the volume control and protected the ear from intolerably loud, unexpected sounds. In the past ten years, however, a further claim of "improved hearing in noise" has appeared in the literature. This claim is accompanied by little, if any, technical information and no experimental evidence, whatsoever. The following abstracts from manufacturers' literature illustrate this claim:

> Automatic control of acoustic gain based on the intensity of environment sounds assured the user of ... better hearing in noise because amplification is automatically reduced as sound input rises (65).

> The low distortion at high input levels makes it possible for patients to hear clearly and comfortably even in noisy group situations (21).

> A powerful eyeglass hearing aid with a special compound volume control circuit ... enables the patient to hear better in noisy situations (21).

> Protection of the ear against over-loud sounds and elimination of the need for fiddling with the volume control in the presence of fluctuating sound levels are important advantages of an AVC circuit. A greater value can be found in the improved hearing of speech in the presence of noise (58).

Despite the real or assumed advantages of compression amplification, the majority of commercially available hearing aids still employ clipping as their method of limiting maximum power output.

The Effects of Limiting on the Intelligibility of Speech in Quiet

A major concern for users of any system involved in the transmission, recording or reproduction of sound is the effect of the system on the intelligibility of speech. The widespread use of peak clippers and compressors in various communication systems has led to investigations of the effects of these instruments on the intelligibility of speech. The following sub-sections are devoted to a discussion of the effects of peak clipping and compression on intelligibility.

The Effects of Peak Clipping on the Intelligibility of Speech

In 1944, Kryter and Stein (29) reported that the intelligibility of speech heard over communication equipment with "limited power capability" could be improved by boosting the level of the weak consonant sounds and limiting the level of the intense vowel sounds. The peak clipper provided a relatively efficient method of limiting while at the same time it boosted the level of the consonants making the information contained in these sounds available to the listener. The authors showed that if the transmitted signal was kept just below the level at which 100 per cent modulation occurred, the consonants averaged only 35 per cent modulation. With 12dB of peak clipping, however, the consonants averaged 70 per cent modulation and with 24dB of clipping the consonants averaged 95 per cent modulation. The additional information provided by modulation of the consonants improved the intelligibility of the transmitted signal over the unclipped condition when the average radiated power was held constant over the two conditions.

Licklider (33) also investigated the effect of peak clipping on intelligibility. He presented monosyllabic words to experienced listeners and found that either symmetrical or asymmetrical peak clipping had minimal effect on the intelligibility of speech. Licklider stated that a signal limited to one-tenth of its original amplitude was 96 per cent intelligible to the trained listener. He concluded that speech which was infinitely clipped, or reduced to a series of rectangular waves was, "surprisingly intelligible."

In 1946, Kryter, Licklider, and Stevens (28) compared the intelligibility of non-limited and clipped speech when the signal peaks were kept below the 100 per cent point of modulation. Monosyllabic speech materials were presented to eight listeners for 0, 6, 12, 18, and 24dB of clipping. The authors found that the intelligibility of the signal was progressively increased as more clipping was utilized because clipping allowed an increase in the average signal level modulating the carrier. Twenty-four decibels of clipping improved the intelligibility of the speech signal by as much as 50 per cent. Although intelligibility was increased, the subjects reported a definite deterioration in the quality of the signal. For 6dB of clipping the signal was described as "essentially normal...effect barely detectable" and the quality judgment was "probably acceptable as of broadcast quality." For 18dB of clipping the signal was described as "sharp", "sandy" and the quality judgment was "fair, usable for military and some commercial communication." For 24dB of clipping the signal was described as coarse, grainy, and unnatural, and the quality was judged as, "poor, but usable if intelligibility is of paramount importance."

In 1948, Licklider, Bindra, and Pollack (34) investigated the intelligibility of clipped speech in quiet for 10 through 100dB of peak clipping in increments of 10dB. The measure of intelligibility was the "per cent word articulation" score for monosyllabic words. The authors demonstrated that the intelligibility of non-limited speech was 100 per cent. For the clipped conditions, intelligibility remained fairly constant for as much as 20dB of clipping. A further increase in the amount of clipping resulted in a reduction of the word-articulation scores. For 50dB of clipping approximately 72 per cent of the words were correctly identified.

This performance was attained by a single person who listened to a limited test vocabulary under many conditions. When the performance of two other subjects who had listened to more difficult materials were compared with that of the first subject at 0dB of clipping and infinite clipping, the scores varied over a range of 20 per cent in the direction of poorer intelligibility.

Because of the differences observed between the results obtained for different sets of materials and among subjects with limited and extensive practice, the authors investigated the effects of these variables on the intelligibility of clipped and non-limited speech. They found a definite learning factor for clipped speech which contributed as much as 23 per cent to the intelligibility scores.

Licklider and Pollack (35) further investigated learning effects in isolation by studying the influence of word sequence and restricted vocabulary on the intelligibility of clipped speech. The authors found that a limited vocabulary and repeated exposure to clipped speech improved the intelligibility scores from 65 per cent for the first tests administered to 90 per cent upon the completion of the experiment. The authors pointed out that a score of 90 per cent was attributable to the diligence of the listeners and the use of a limited sample of words. These scores are as much as 22 per cent better than those obtained by Licklider, Bindra, and Pollack (34) for the initial listening conditions and 30 per cent better for the scores obtained upon the completion of the tests.

Licklider and Pollack (35) investigated another variable which they claimed influenced the intelligibility of clipped speech. This variable was the frequency response of the circuits existing prior to and following the peak clipper. The authors compared the effects of infinite peak clipping, integration (a 6dB fall per octave) and differentiation (a 6dB rise per octave) in isolation and in various combinations.

The authors found that differentiation and integration in isolation had a minimal effect on the intelligibility of speech. They observed that the effect of peak clipping on speech intelligibility depended on the frequency response of the circuit that preceded it. If a circuit employing a 6dB rise per octave preceded the clipper, the detrimental effect of clipping was overcome and intelligibility reached as high as 95 per cent. If a circuit employing a 6dB fall per octave preceded the clipper, the intelligibility of the speech signal deteriorated to as low as 15 to 24 per cent.

Licklider and Pollack concluded that tests with single distortions and combinations of distortions indicated that in the absence of frequency distortion, infinitely clipped speech was of poor quality but moderate intelligibility (50 to 90 per cent) depending on the listener's skill and familiarity with the test words.

The Effects of Compression Amplification on the Intelligibility of Speech

In 1952, Edgardh (17) attempted to construct a new type of spectrograph which could be utilized for the analysis of the dynamic acoustics of linquistic sounds. The author used extreme limitation, achieved by compression amplification, for the purpose of dynamically equalizing vowels and consonants to the same sound pressure levels. He found that in equalizing the speech elements the character of the sound was somewhat altered. Breath gasps were more obvious and a certain sibilance appeared. Despite these minor disadvantages, Edgardh stated that there was "no such distortion of speech as to affect adversely its comprehension... either in male or female voices."

In 1953, Parker (44) put forth the hypothesis that certain components of speech tended to "fatigue" the ear and that when these elements of speech were removed the intelligibility of the speech stimulus was minimally affected. He held that the advantage resulting from the reduction of the "fatiguing" elements of speech would outweigh any detrimental effect on the intelligibility of speech caused by the signal processing with the result that an individual with an inner-ear hearing loss would find the altered speech more understandable.

Parker reduced the strongest speech sounds by high-pass filtering, compression amplification, and speech-time fractionating. After compressing the speech signal three times, the resultant tapes were extremely noisy since full amplification was afforded the weak background noise. The between-the-signal noise was removed by recording the signal through a voice operated relay which was adjusted to differentiate between the speech signal and the noise between words. The final result was a signal with a dynamic range of less than 10dB.

Parker established articulation functions for hard-of-hearing subjects by presenting word lists at sensation levels of 6, 16, 26, and 36dB. The responses of the subjects to compressed speech were extremely varied. The general trend was that compression of the speech signal resulted in improved intelligibility primarily at the lower sensation levels. Most of the subjects reached a point of maximum performance at a lower sensation level for compressed speech than for uncompressed speech. Parker speculated that his subjects experienced a temporary threshold shift from the high presentation level of speech required for hard-of-hearing individuals and that this shift in threshold was sufficient to impair intelligibility. By compressing the speech signal, the weaker portions of speech were amplified sufficiently to contribute to speech intelligibility whereas, without compression they would have been below the threshold of audibility for the hearing-impaired individual. At the same time the maximum signal peaks were compressed, which, according to Parker, prevented fatigue of the ear.

In 1962, Lynn (38) investigated the parameters of the attack and release times of compressor amplifiers as they affected the intelligibility of speech for hard-of-hearing listeners. His hard-of-hearing groups consisted of individuals exhibiting hearing losses due to otosclerosis, labyrinthine hydrops, and presbycusis. Lynn used a peak limiter constructed from hearing aid components. Nine different sets of time constants were available with attack times ranging from 5 to 85 msec and release times ranging from 30 to 1,200 msec.

Lynn established speech reception thresholds and obtained discrimination scores for nine conditions of compressed speech and one condition of uncompressed speech. The major findings of the study are summarized as follows:

1. For every group, speech reception thresholds for each compressed-speech condition were better than for the uncompressed-speech condition.

2. Time constants had an important influence on the discrimination ability of hard-of-hearing subjects as measured with phonetically-balanced material.

3. Hydrops and presbycusis cases showed slightly improved discrimination ability with the shorter time constant conditions.

4. For longer time constants (20/500, 70/400, and 85/1,200) discrimination scores were essentially equivalent or inferior to the uncompressed-speech condition.

5. Otosclerotics showed no exceptional difference in discrimination ability with or without compression except for the longest time constants (85/1,200) where discrimination was definitely poorer.

Lynn suggested that the use of compression circuitry in hearing aids with properly chosen characteristics should improve the intelligibility of speech by possibly 16 to 23 per cent for individuals with hearing losses caused by labyrinthine hydrops and presbycusis.

A year later, Lynn and Carhart (39) in summarizing the clinical implications of Lynn's findings, stated "that the value which a hard-of-hearing person will find in compression amplification will depend on several factors...(a) the type of compression system being used, (b) the time constants of the system, (c) the nature of the user's hearing impairment, and (d) the levels and varieties of sounds which constitute his acoustic environment."

Caraway (7), in 1964, investigated another parameter of compression amplification, that of compression ratio. She hypothesized that, "...a reduction in the dynamic range of speech as achieved by a compression amplifier with a 2:1 and 3:1 compression ratio, would increase the intelligibility of speech over a linear system when the peak powers of the acoustic signal were held constant."

To test this hypothesis, Caraway utilized normal-hearing subjects and three groups of hard-of-hearing subjects. The losses of the hearing-impaired groups were due to labyrinthine hydrops, labyrinthine otosclerosis, and presbycusis. Speech reception thresholds and articulation functions were obtained for all subjects for the three conditions of amplification.

The major findings of Caraway's study may be summarized as follows:

1. The dynamic range reduction offered by the two compressed-speech conditions did not create any distinguishable changes in the speech reception thresholds as compared to the uncompressed-speech reception thresholds for all four groups.

2. The normal hearing and labyrinthine-hydrops groups showed superior discrimination scores for the compressed-speech condition at the 8dB sensation level as compared to the uncompressed-speech condition. The labyrinthine otosclerotics performed equally well under all amplification conditions. The presbycusis group performed similarly except at 24dB SL. At this level, they did poorer under the 3:1 condition than under the 2:1 condition.

3. The discrimination scores of the hearing-impaired groups were, in general, the same under all conditions of amplification.

On the basis of these findings Caraway concluded that the reduction of the dynamic range of speech did not greatly increase the intelligibility of the signal for the sensori-neural hearing loss groups studied. Indeed, the greatest improvement in intelligibility for compressed speech was demonstrated by the normal-hearing subjects, and this improvement was confined to a single presentation level.

Comparative Studies on the Relative Intelligibility of Clipped and Compressed Speech in Quiet

In 1947, Davis, Stevens, and Nichols (15) published a paper on the objectives of hearing aid design in which they discussed the relative effects of clipping and compression on the intelligibility of speech. The authors reported on "A Master Hearing Aid" which had been built several years earlier at the Electro-Acoustic Laboratory at Harvard University (14). This aid was capable of achieving five different frequency response patterns and several levels of maximum acoustic output. Either peak clipping or compression could be used to limit acoustic power. Davis and his associates used a series of PB-50 speech discrimination tests to compare the effects of clipping and compression. They found that the average discrimination scores made by three normal-hearing subjects and six hard-of-hearing subjects were higher when using thirty decibels of compression for all five frequency response patterns than for the same patterns presented under conditions employing thirty decibels of clipping.

The major conclusion of this study in regard to limiting acoustic power output was:

> Of the available devices the simplest is the peak clipper. Properly adjusted peak-clipping protects the ear from discomfort and pain while allowing a predetermined maximum amplitude of signal to be delivered. The intelligibility of speech is not seriously reduced by as much as 12dB of peak-clipping. "Compression amplification" produces less amplitude distortion than simple peak clipping and if an effective compressor can be built into a wearable hearing aid it may provide the ideal means of limiting the maximum acoustic output.

In 1948, Hudgins, Marquis, Nichols, Peterson, and Ross (25) built a wearable hearing aid which employed compression as the method of limiting maximum acoustic power output. The authors utilized characteristics which previous work at Harvard's Psycho-Acoustic Laboratory had proved to be the most advantageous for hard-of-hearing individuals. The performance of the experimental aid was compared to that of the original Master Hearing Aid constructed at Harvard University and two commercially available hearing aids. Speech discrimination tests were administered through each instrument to six hard-of-hearing subjects. The results showed that the Master Hearing Aid performed as well as, or better than, the other hearing aids. The experimental aid performed second only to the Master Hearing Aid. The authors concluded that the results:

> ...indicate clearly both the feasibility and the desirability of limiting the output of hearing aids by means of compression amplification. Furthermore, it was demonstrated that limiting the power by means of compression amplification not only protected the ear, but at the same time reduced distortion to a minimum, thus maintaining in most cases the maximum level of performance over a wider range of speech-input levels.

The Effects of Limiting on the Intelligibility of Speech in Noise

The effects of limiting on the intelligibility of speech in the presence of noise have been investigated under two basic conditions:

(1) when the listener is located in or is surrounded by background noise and (2) when the talker is located in or is surrounded by background noise. It is important to note that in the first condition only the speech signal is acted upon by the limiting system and that under the second condition both speech and noise are processed by the limiting system.

The Intelligibility of Limited Speech When
the Listener Is in Noise

In 1946, Kryter, Licklider, and Stevens (28) investigated the effect of peak clipping on the intelligiblity of speech in noise. They utilized 6, 12, 18, and 24dB of peak clipping in two different situations. The first condition occurred when the listener was in 120dB SPL of simulated airplane noise and the second condition occurred when the listener was in 120dB SPL of simulated atmospheric static. Eight subjects were asked to identify monosyllabic words read by an experienced talker.

Under both conditions of noise the improvement in intelligibility afforded by peak clipping was as great as 50 per cent compared to non-limited speech. This improvement was attributed to the boosting of the average signal level reaching the subjects as a result of the use of peak clipping.

In 1948, Licklider, Bindra, and Pollack (34) further investigated the effects of peak clipping on speech intelligibility when the listener was located in a noisy environment. From their investigation the authors concluded that the square waves of infinitely clipped speech resulted in improved intelligibility compared to that offered by the non-limited or irregular waves of speech. They claimed that the superiority of the square wave was due to an improvement in the "power per unit peak amplitude." Since the consonant sounds were boosted and the more intense vowels were limited, all portions of the speech signal remained audible in high levels of noise. During the production of non-limited speech waves, the consonants, which were extremely important for speech intelligibility, were masked by the background noise.

In 1960, Kretsinger and Young (27) compared the effects of peak clipping and compression on the intelligibility of speech mixed with noise. The authors suggested that the conventional peak clipper produced harmonic and intermodulation distortion, and that such distortion masked speech, reducing its intelligibility in much the same way as does ambient noise. They stated that any system which could produce the same consonant-to-vowel ratio as a speech clipper without its associated distortion should yield more intelligible speech where noise was a masking factor.

To test their hypothesis, Kretsinger and Young presented speech discrimination tests to thirty normal-hearing subjects. The PB-50 word lists were submitted to 10 and 20dB of clipping and compression. The clipped-speech signal was also filtered in order to suppress "unwanted harmonics." The compressed speech signal was mixed with white noise maintained at 3dB below the level of the speech and dubbed onto another tape. The same procedure was repeated for the clipped speech signal.

Kretsinger and Young found that their subjects obtained significantly higher intelligibility scores for compressed speech than for clipped speech in the presence of noise. The authors suggested that an area for further study was an investigation of the relationships between amounts of compression and various signal-to-noise ratios.

The Intelligibility of Limited Speech When the Talker Is in Noise

In 1944, Licklider (33) investigated the effects of peak clipping on the intelligibility of speech when noise entered the speaker's microphone at the same time as the speech signal. The author suggested that the effect of peak clipping was deleterious to the intelligibility of speech produced in the presence of noise. To test this hypothesis Licklider compared intelligibility scores obtained by normal-hearing subjects who listened to monosyllabic words presented through two different microphones. When a microphone that was not sensitive to background noise was used, the speech discrimination scores were only slightly poorer than those obtained in quiet. When another microphone which picked up the background noise was used, the scores were reduced. In general, 12dB of clipping resulted in a 10 per cent reduction in the intelligibility of speech. For 24dB of peak clipping the intelligibility scores were reduced an additional 14 per cent. The subjects scored an average of only 56 per cent when noise was picked up by the talker's microphone.

Licklider concluded that, not only is the type of microphone important in determining the effects of peak clipping on the intelligibility of speech, but the type of background noise is also a major concern. He suggested that low-frequency noise with a low-peak factor reduced the tolerance for peak clipping.

Kryter, Licklider, and Stevens (28) also investigated the effects of peak clipping on speech intelligibility when the noise was located at the signal's source. They found that intelligibility scores became poorer as more clipping was used. The authors pointed out that when speech and noise are mixed prior to the peak clipper that "speech tends to ride ride on the noise" carrying the peaks of the signal beyond the threshold of clipping. Besides the loss of information incurred by limiting the signal peaks, what was left of the speech signal appeared to be masked by the intermodulation products. The authors concluded that, "Intermodulation, then, makes inadvisable the use of extreme peak clipping when the microphone is exposed to intense noise." They added, "Since the optimum amount of peak clipping is determined by the noise-rejection characteristics of the microphone, by the spectrum and intensity of the ambient noise, by the voice level used by the talker, and perhaps by other factors, it is not possible to recommend a single amount of clipping for use under all conditions."

At present, there does not appear to be any literature available in which the effect of compression amplification on the intelligibility of speech when noise enters the system has been investigated. There are, however, several references in which incidental speculation is made regarding the effects of compression and clipping on signal-to-noise ratio. These scattered opinions are discussed in the following section.

The Effects of Limiting on the Signal-to-Noise Ratio

Silverman, Taylor, and Davis (55) were among the first to suggest that the use of compression amplification in hearing aids improved the intelligibility of speech in the presence of noise. The authors illustrated the relationship existing between a speech signal and a background noise before and after entering hearing aids employing compression and clipping as methods of limiting maximum acoustic output. They suggested that compression amplification offered better hearing in noise by preserving a more favorable signal-to-noise ratio than that afforded by peak clipping. No experimental evidence was given to support the authors' claim.

Rutherford (51) also held that clipping would obliterate the peaks of speech which contributed to intelligibility, resulting in a decrease in the speech-to-background noise ratio. He believed that a compressor would preserve the same signal-to-noise ratio that existed originally.

In sharp contrast, Kretsinger and Young (27) predicted that "a unique masking problem" would occur if noise were allowed to enter a compressor along with the speech signal. They reasoned as follows:

> Noise entering prior to compression (particularly circuit noise with amplitudes similar to that of the weak consonants) will receive the same full amplification accorded the consonants. Thus, while the consonant-to-vowel ratio will be improved, the *vowel-to-noise ratio will suffer. To prevent the latter condition* from giving rise to a masking problem of its own, it is imperative that noise be minimized in the preamplifier....Acoustic background noise at the microphone site must be controlled.

Krebs (26) commended AVC instruments because of their ability to limit output with minimal harmonic distortion. At the same time, he presumed that under certain conditions AVC hearing aids could prove to be disadvantageous. Krebs stated that:

> Any sound which causes the instrument to compress will cause any other sound being handled by the instrument at the same instant, to be reduced in level by the amount that the instrument is in compression. If the softer sound is the background noise and the louder sound speech, an advantage over peak clipping has been gained. If, however, the louder sound is the background *noise, the secondary speech signal is certainly going to be at a* disadvantage in the compression instrument as compared to a conventional peak clipping instrument.

Ling (36), in discussing his auditory approach to the education of deaf children, stated that, "background noise often causes problems for children with hearing aids....To overcome this problem, we use hearing aids without AVC and systematically employ a low gain and a high acoustic input to differentiate between foreground (signal) and background (noise)."

Application to the Present Investigation

It is apparent that the use of both peak clipping and compression amplification has proved to be advantageous in the broadcasting, recording, communication, and sound reinforcement industries, and to an extent, in the hearing aid industry. There appears, however, to be some confusion in regard to the effectiveness of these two methods of limiting when they are used under circumstances in which noise can enter the system along with speech. Perhpas part of this confusion stems from the fact that there are three distinct locations in a communication system in which the introduction of noise becomes a problem.

Any communication system may be conceived of as being composed of a signal source, a transmission channel or medium, and a receiver. The introduction of noise at any one of these points can create a serious problem in communication.

In broadcasting and recording, noise at the signal source rarely becomes a problem because of the careful attention paid to the acoustical treatment of the studio. It is this very situation, however, which is directly analagous to the circumstances under which a hearing aid is commonly used, that in which environmental noise is free to enter the system along with speech.

Noise in the transmission channel, however, is a very common problem because of equipment noise, atmospheric conditions, tape hiss, or disk noise. This problem can be solved by boosting the level of the weaker portions of a signal at an earlier stage so that these sounds are not lost in, or masked by, the background noise. Boosting the over all level of the signal is impractical because of the danger of overmodulating the system. Clipping the peaks of the signal allows the average signal level to be increased, with the result that the weaker sounds are boosted while at the same time signal peaks are limited.

In military communication systems, signal power is limited, intelligibility is paramount, and the quality of the transmission is not important. Under these circumstances, peak clipping may be used to increase the intelligibility of the transmitted signal by as much as 50 per cent by allowing the consonants to modulate the carrier signal fully.

Although clipping may solve one problem, at the same time it gives rise to another in that the signal suffers in quality because of the distortion associated with clipping. In commercial broadcasting and recording, of course, poor signal quality is not tolerable.

Because compression permits effective limiting without distortion of the signal waveform it has become an almost universally accepted method of limiting. Moreover, in certain applications it can be used to improve the signal-to-noise ratio existing in the transmission channel or the recording medium.

The third possible locus of noise is at the signal's destination. Sound reproduced in a theater, machine shop, business office, or transportation terminal often is masked by ambient noise. Merely boosting the level of the signal above the existing noise is not practical since the level which would be required to achieve audibility of the weak sounds would be intolerable to the listener's ear.

Again, peak clipping may solve this problem by packaging sound in a more efficient waveform in which the levels of the consonants and vowels are equalized. The result is that the consonants maintain audibility while at the same time the more intense vowel sounds are limited. Because compression amplification can perform this task without the distortion associated with clipping, it has become the preferred method of limiting.

The advantages of peak clipping and compression amplification in the broadcasting, recording, communication, and sound reinforcement industries have been supported by research concerning the effects of these methods of limiting on the intelligibility of speech. Licklider, Bindra, and Pollack (34), for example, demonstrated that "when equated in peak amplitude with normal speech-waves and heard in the presence of intense noise, the square waves of infinitely clipped speech are even more audible and more intelligible than the irregular waves of normal speech." Kretsinger and Young (27) suggested that any system which could produce the same consonant-to-vowel ratio as a speech clipper without its associated distortion should yield more intelligible speech where noise was a masking factor and the authors demonstrated experimentally that the compression amplifier was, indeed, such a system.

In recent years, the hearing aid industry has recognized some advantages of peak clipping and compression amplification, and, certainly, some of the advantages derived through the use of limiting in other communication systems are also shared by the user of a hearing aid. Both peak clipping and signal compression reduce the need for constant manual adjustment of the volume control of a hearing aid and prevent the reproduction of sudden, intolerably loud sounds. It remains to be demonstrated, however, whether or not the improvement in signal-to-noise ratio made possible by peak clipping and compression in other applications can be realized with hearing aids. The manufacturer's literature on hearing aids which employ compression as their method of limiting maximum acoustic power output suggest that the use of these instruments "improves hearing in noise," but no compelling evidence in support of this claim has been presented.

The circumstances under which a hearing aid is commonly used are not analagous to the circumstances in which the use of compression has proved to be advantageous in the broadcasting and recording industries. In hearing aid use, the problem is unique in that environmental noise is free to enter the system at the source along with speech. The effects of limiting on the intelligibility of speech under these particular circumstances have not been widely investigated and most of the few opinions which have been voiced in regard to this point appear to be based on speculation rather than the results of experimentation.

Kryter, Licklider, and Stevens (28) indicated that the use of peak clipping when noise enters the system was deleterious to the intelligibility of speech because of the intermodulation of speech and noise. The speech signal appeared to "ride" the crest of the noise, so that the signal peaks which contributed to the intelligibility of the signal were clipped, and what was left of the signal was masked by the intermodulation products.

Silverman, Taylor, and Davis (55) suggested that compression amplification offered better hearing in noise by preserving a more favorable signal-to-noise ratio than that afforded under conditions of peak clipping. Kretsinger and Young (27) predicted that when noise entered prior to a compressor "a unique masking problem" occurred. Rutherford (51) held, however, that the speech-to-background noise ratio at the output of the compressor would remain the same as that existing in the environment, but that the signal-to-noise ratio at the output of the clipper would be degraded. Krebs (26) also suggested that compression would have no effect on the signal-to-noise relationship, but he went on to comment that if the background noise was softer than the speech signal compression offered an advantage relative to peak clipping, but if the background noise was the louder signal he predicted that clipping would offer an advantage over compression. Ling (36) preferred conventional aids to AVC hearing aids in training deaf children to differentiate between background noise and speech.

In view of the conflict of opinion, the lack of available evidence in the matter, and the importance of the implications of the question of whether or not compressed or clipped speech is more intelligible in the presence of noise to hearing aid use, the present investigation was designed to explore the comparative intelligibility of compressed, clipped and non-limited speech in quiet and for three signal-to-noise ratios.

The following chapter is devoted to a description of the subjects, test materials, apparatus, and procedures employed in the conduct of the experiment.

CHAPTER III

SUBJECTS, TEST MATERIALS, INSTRUMENTATION, AND PROCEDURES

Introduction

The present investigation was designed to compare the relative intelligibility of compressed, clipped and non-limited speeoh in quiet and when noise entered the communication system at the same time and place as the speech signal. Recordings were made of three lists of CNC words of demonstrated equivalence and of cafeteria noise. Upon playback of these recordings to normal-hearing subjects, the speech and noise were combined at the source to provide the desired signal-to-noise ratios. The combined signal was processed through three separate channels. Three subjects were tested at a time with one listening to compressed speech, another listening to clipped speech, and the third listening to non-limited speech. The order of presentation of the word lists and the order of occurrence of the experimental conditions were counterbalanced. Four randomizations of each list were presented at -5, 0, and +5dB signal-to-noise ratios and without noise, in that order, under each condition of amplification. The subjects' task involved the identification of single monosyllabic words from a fifty-word list presented at a sensation level of 40dB. The following sections of this chapter are devoted to a detailed description of the subjects, test materials, apparatus, and procedures employed in the investigation.

Subjects

Eighteen adults between the ages of 22 and 30 served as subjects in the present investigation. Seven were female and eleven were male. Audiometric screening of the subjects was accomplished with a pure-tone audiometer (Beltone, 15CX). The output of the audiometer was calibrated in accordance with the 1969 ANSI Standard for pure-tone audiometers. Each of the subjects met the following criteria:

1. He had a threshold for pure tones of 20dB or better re: ANSI 1969 Norms at octave intervals between 250 and 4,000 Hz.

2. He had a negative history of noise exposure and ear pathology.

Acoustic Environment

The familiarization of the subjects with the test materials, the establishment of speech reception thresholds, and the collection of the experimental data were all accomplished in a two-room acoustically-treated audiometric test suite located at the Speech and Hearing Center at the University of Oklahoma Medical Center. The suite consisted of an examiner's room and an examinee's room. The examiner's room contained the experimental equipment which was necessary for the conduct of the experiment. The examinee's room was equipped with three desks, three headsets, and a VU meter (Weston 802). The desks provided a flat surface so that each subject could record his responses on paper. The VU meter provided a means for the subjects to monitor the presentation of each test word visually. Verbal communication was maintained between the rooms by means of a talk-back system.

Test Materials

Two sets of speech materials were utilized in the conduct of the experiment. Spondee words were used for the establishment of each subject's speech reception threshold and monosyllabic words were used to assess intelligibility. A recording of cafeteria noise served as a competing stimulus. Descriptions of the two sets of materials and the noise appear in the following sub-sections.

Spondaic Words

The 36 spondee words comprising Auditory Test W-2 developed at the Central Institute of the Deaf were recorded at the University of Oklahoma Speech and Hearing Center by Sommerville (57). The words were spoken by a male talker and were recorded at a constant level. A copy of this recording was used to establish speech reception thresholds in the present investigation. The 36 words appear in alphabetical order in Appendix A.

CNC Words

In compiling a body of words for a speech discrimination test, Lehiste and Peterson (32) selected 1,263 monosyllabic words listed by Thorndike and Lorge (61) as occurring at least once per million words. Lehiste and Peterson referred to these words as CNC (Consonant-Nucleus-Consonant) words since each word consisted of an initial consonant, a vowel nucleus, and a final consonant. The authors selected 500 words from the 1,263. They divided these into ten lists of fifty words per list. The frequency of occurrence of each initial, medial, and final phoneme in each of the ten lists was representative of that observed in the original pool of 1,263 words.

Carhart, Tillman, and Wilber (9) utilized 100 of Lehiste and Peterson's CNC words in the development of N.U. Auditory Test No. 4, a test of speech discrimination. Carhart and his associates selected two groups of fifty words which preserved the phonemic balance of Lehiste and Peterson's original 1,263 words. The authors recorded six randomizations of each of the two word lists. The recordings of these test materials were found to offer good test reliability and inter-list equivalence, but the format was unduly restrictive. In projects which involved an extensive testing of an individual's speech discrimination ability the experimenters found it necessary to use each of the two lists of words repeatedly. This contributed to a learning effect that influenced the responses of the subjects and biased the results of the test.

Tillman and Carhart (62) expanded Test No. 4 to solve this problem. The resulting test, N.U. Auditory Test No. 6, consisted of four lists of fifty words each. Each of the four lists met the criteria for phonemic balance advocated by Lehiste and Peterson. All but fifteen of the words comprising Test No. 6 were selected from Lehiste and Peterson's list of 500 words. The remaining words were taken from the original pool of 1,263 words. Tillman and Carhart recorded four randomizations of each word list.

Sommerville (57) recorded the N.U. Auditory Test No. 6 word lists in order to provide the Speech and Hearing Center of the University of Oklahoma Medical Center with taped speech materials which could be utilized for research and clinical purposes. The words were spoken by a male talker using general American dialect. Sommerville experimentally verified the interchangeability of the new recordings. Lists II, III, and IV of these recordings were used in the conduct of the present investigation. The 150 words comprising these three lists appear in alphabetical order in Appendix B.

The Noise

Cafeteria noise was arbitrarily selected as the competing auditory stimulus to be used in the present investigation. The noise presented to the subjects consisted of a four and one-half minute recording made during a routine lunch hour at the staff dining room of the University of Oklahoma Hospital, Oklahoma City, Oklahoma. The recorded material consisted primarily of the babble of a multitude of voices accompanied by the clanking of china and utensils.

Recording Apparatus and Procedures

The experimental procedure employed in the present investigation called for the presentation of three different lists of materials at four different signal-to-noise ratios under three conditions of amplification. The use of individual recordings for each of the experimental conditions would have required 36 separate tapes. It is difficult to maintain uniformity among such a large number of recordings. It was also considered desirable to insure that the noise was identical over all three experimental conditions. Consequently, it was decided to develop a more elaborate apparatus which would allow the simultaneous processing of a single set of master recordings through three separate channels, one for each of the three principle experimental conditions.

The Recording Apparatus

A simplified block diagram of the equipment used for the recording of the experimental tapes appears in Figure 2. One channel of a dual-channel tape recorder (Ampex 354), designated TAPE RECORDER 1 in the block diagram, was used for the reproduction of the speech materials.

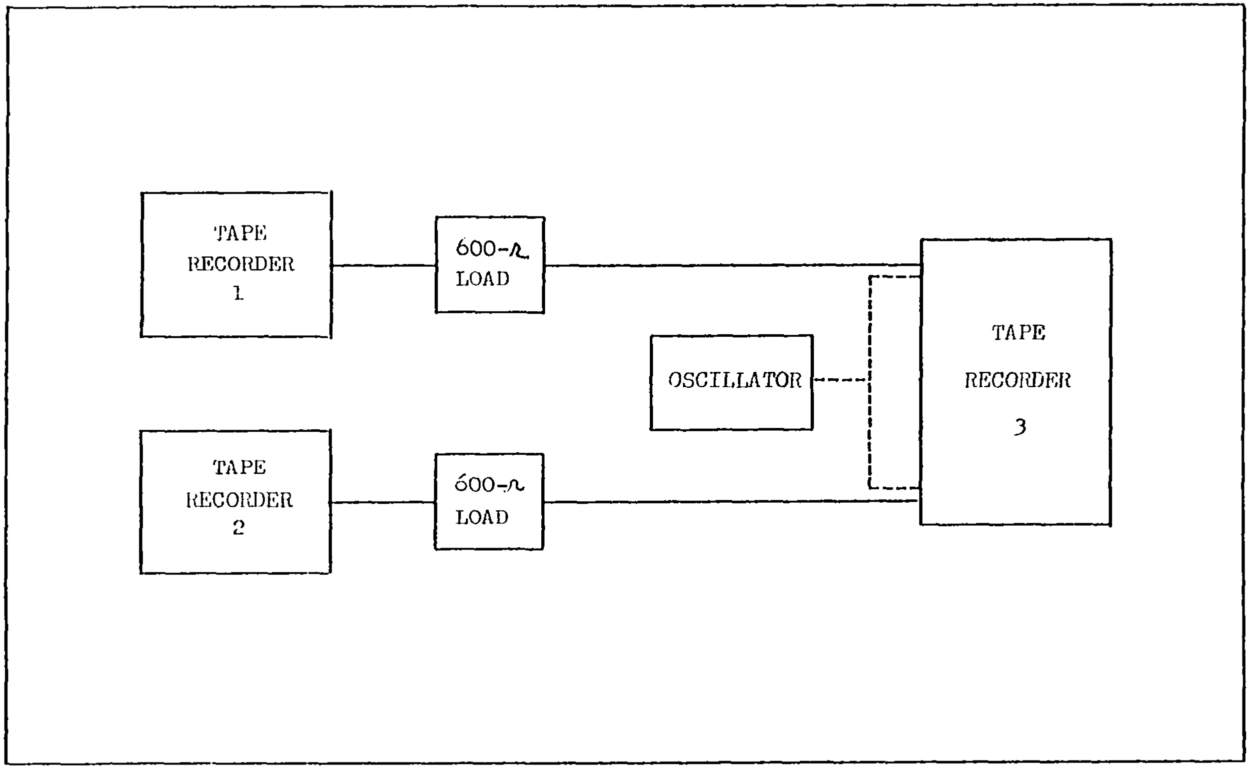


Figure 2--A simplified block diagram of the equipment used for the recording of the experimental tapes.

A single-channel tape recorder (Ampex 601) designated TAPE RECORDER 2, reproduced the noise. The performance of both tape recorders was evaluated with the equipment and procedures recommended by the manufacturer of the units. Tape recorder 1 used for the reproduction of the speech material performed within its manufacturer's specifications in the playback mode. Tape recorder 2 used for the reproduction of noise performed within its manufacturer's specifications for the playback mode in all but one respect. The unit's signal-to-noise ratio fell slightly short of that specified, but this was considered acceptable for the purposes of the present investigation because the material reproduced on this recorder was noise.

The output of each tape recorder was loaded with a 600-ohm resistor and paralleled by the high-impedance input of one channel of a dual-channel tape recorder (Ampex 601-2) designated as TAPE RECORDER 3 in the diagram. The performance of this recorder was evaluated in both the playback and record modes. Both channels of the recorder were adjusted to be within the specifications published by the manufacturer.

A 1,000-Hz signal generated by an oscillator (Hewlett Packard 201CR), designated as OSCILLATOR in the diagram, provided a calibration tone which was recorded at the beginning of each word list.

The Recording Procedure

The investigation required the recording of thirteen master tapes. Tape 1 was comprised of a calibration tone and a spondee word list. Tapes 2 through 13 were each composed of a calibration tone and a CNC word list on one track and a calibration tone and noise on the other track. The level of the calibration tone was set to equal the average level of the peaks of each set of materials as measured with a standard VU meter.

The original speech and noise tapes were placed on the playback tape recorders during the recording of the master tapes. Immediately prior to recording each word list on tape recorder 3, the oscillator was substituted for tape recorders 1 and 2. The level of the 1,000-Hz tone produced by the oscillator was adjusted to a VU meter reading of -10 on both channels of tape recorder 3. Upon placing tape recorder 3 in the record mode a tone of thirty seconds duration was recorded on both channels. The oscillator was then removed from the circuit and tape recorders 1 and 2 were connected.

Upon placing tape recorder 3 in the record mode, tape recorder 1, used for the reproduction of the speech materials, was started in the playback mode. Tape recorder 2 was started in the playback mode immediately following the introduction of the word list by the talker. All three of the tape recorders were stopped upon the termination of the word list. This procedure was repeated for each word list after rewinding the noise tape.

Following the recording of each CNC word list and the spondee word list, the tapes were spliced and placed on four separate reels. Reel 1 consisted of the spondee word list and its calibration tone. Reels 2, 3, and 4 consisted of four randomizations each of Lists II, III, and IV, one list to each reel, of copies of Somerville's recordings of the N.U. Auditory Test No. 6 speech materials on one track of the tape and a recording of noise on the other track. The four randomizations were designated A, B, C, and D.

The Experimental Apparatus

The experimental design called for an apparatus which allowed the simultaneous processing of speech and noise recordings through three separate channels. One channel provided a non-limited signal, a second channel provided a compressed signal and a third channel provided a clipped signal. The output of each channel was presented at a 40dB sensation level to one of three subjects who were tested simultaneously. A simplified block diagram of the equipment utilized for the playback of the experimental tapes appears in Figure 3. In order to simplify discussion, the equipment will be described in the following order: the portions of the apparatus common to all three channels, the non-limiting channel, the compressed channel, and the clipping channel.

Apparatus Common to All Three Channels

The same dual-channel tape recorder (Ampex 601-2) on which the master tapes were recorded was also used for the reproduction of the experimental tapes. Channel I of the recorder was used for the reproduction of the speech materials and Channel II was used for the reproduction of the noise. A VU Meter (Weston 802) was bridged across the output of Channel I. The meter was placed in view of the subjects to permit visual monitoring of the occurrence of each test word. The output of each channel of the recorder was routed to an attenuator (Hewlett Packard 350 D). These two attenuators, labeled ATTEN. 1 and ATTEN. 2 in the diagram, provided independent control of the output level of each channel of the tape recorder. The attenuated signals were then mixed in a resistive network (Daven Type 1130-21), designated as MIX in the diagram, the output of which was loaded with a 600-ohm resistor. The input of an

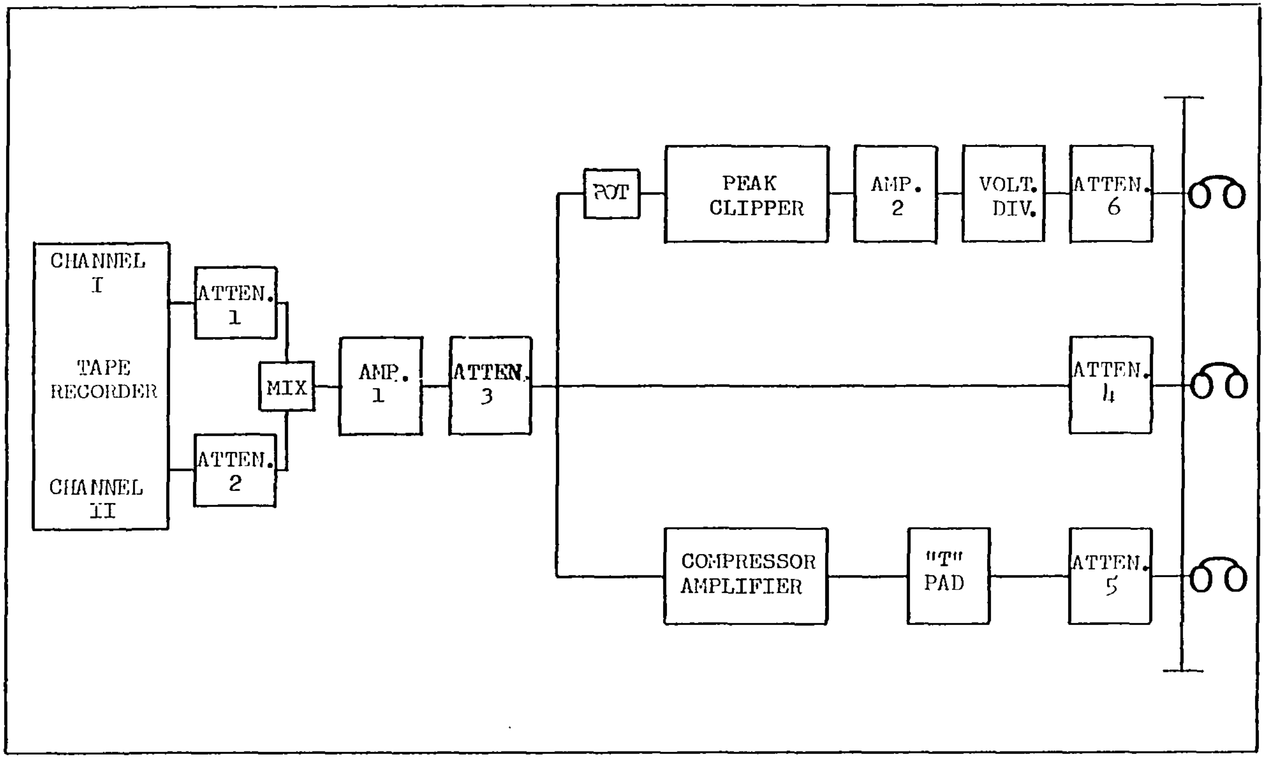


Figure 3--A simplified block diagram of the equipment used for the playback of the experimental tapes.

amplifier (Altec 436), designated as AMP. 1 in the diagram, was paralleled across the load. The amplified signal was channeled to a third attenuator (Hewlett Packard 350AR), labeled as ATTEN. 3 in the diagram. The combined signal was then routed through three separate channels which provided the non-limited, compressed and clipped characteristics.

The Non-Limiting Channel

The non-limiting channel consisted of the apparatus common to all three channels and an attenuator (Hewlett Packard 350 D), designated as ATTEN. 4, which was terminated by an earphone (Permoflux PDR-600). The only active network in this channel was the booster amplifier common to all three channels. The amplifier was operated in the linear portion of its input-output function in order to insure that the amplified signal was not limited.

The Compressor Channel

The output of attenuator 3 of the common portion of the apparatus was bridged by the high-impedance input of a compressor amplifier (Altec 436) designated as COMPRESSOR AMPLIFIER in the diagram. The output of the compressor was routed to a variable "T" Pad (Mallory T600), designated as "T" PAD in the diagram, which was used to equate the level of the output signal to that of the non-limited signal. The adjusted signal was routed to an attenuator (Hewlett Packard 350 D), designated as ATTEN. 5, and then to an earphone (Permoflux PDR-600).

The most important component of the compressor channel was the compressor amplifier. The Altec 436 Compressor Amplifier selected for use in the experiment is an automatic volume control amplifier, the output level of which is limited only after a pre-determined threshold has been exceeded. The amplifier maintains a relatively constant ratio between the input and output signal levels throughout the region of compression. The compression ratio can be adjusted to nominal values ranging from 2:1 to 4:1. The time constants are flexible in that the release time may be varied from .33 to 1.33 seconds while the attack time remains fixed at approximately 50 msec. The threshold of compression, the point at which the limiting action is initiated, is somewhat dependent on the compression ratio selected.

It was arbitrarily decided, for the purpose of this investigation, to select time constants representative of those encountered in currently available automatic-volume-control hearing aids. The time constants of seven such instruments served as a reference for this study. A list of these aids and their time constants can be found in Appendix C. The ranges of the time constants for the seven aids were from 3.5 to 111 msec attack time and from 25 to 500 msec release time. The time constants of 50 msec attack time and 300 msec release time provided by the specific 436 Compressor Amplifier used in this study fell roughly midway within these ranges and were considered acceptable for the purposes of the investigation.

The AVC hearing aids sampled employed compression circuits which were characterized by curvilinear input-output functions varying from approximately a two-to-one compression ratio for low-level input signals to about a five-to-one compression ratio for high-level signals. The 3.3-to-one compression ratio provided by the Altec 436 Compressor Amplifier was accepted since this value lay approximately midway within the range of compression ratios of the hearing aids sampled.

The Clipper Channel

The output of attenuator 3 of the common portion of the apparatus was also paralleled by a 25,000-ohm potentiometer which served as a voltage divider. This potentiometer, designated as POT in the diagram, was used to adjust the voltage delivered to the clipper. The peak clipper was comprised of a voltmeter (Ballantine 300) designated as PEAK CLIPPER and an amplifier (Altec 436), designated as AMP.2, used as an impedance converter. The output of the clipper was terminated in a resistive voltage dividing network, labeled VOLT. DIVIDER, which was used to attenuate the signal delivered to the impedance converter. The output of the impedance converter drove an attenuator (Hewlett Packard 350 D), designated as ATTEN. 6, and its terminating earphone (Permoflux PDR-600). The gain control on the amplifier was used to equate the output level of this channel to that of the other two channels.

The Evaluation of the Experimental Apparatus

It was necessary to evaluate the performance of each piece of equipment utilized in the conduct of the experiment prior to the playback of the experimental tapes. The evaluation procedures and the results obtained for the equipment in each of the three channels will be discussed in the following sub-sections.

The Non-Limiting Channel

The performance of the dual-channel tape recorder (Ampex 601-2) was evaluated in both the playback and record modes with the equipment and procedures recommended by the manufacturer of the unit. Both channels of the tape recorder performed within the specifications published by the manufacturer.

Attenuators 1, 2, and 3 were evaluated and found to be linear within specifications for both the unit and decade steps of the dials.

The frequency response and harmonic distortion of the booster amplifier (Altec 436) were measured in the uncompressed mode. In this mode the amplifier performed as a conventional amplifier exhibiting linear gain characteristics. The harmonic distortion amounted to less than 1 per cent.

Frequency response. A frequency response measurement was obtained at two levels. The output voltage generated by a 1,000-Hz tone was adjusted to read 1 volt RMS across a 600-ohm resistive load and the output voltages at frequencies from 50 to 10,000 Hz were compared to the reference voltage. The frequency response was found to be flat within ± 1dB of the reference voltage from 50 to 10,000 Hz. The measurement was repeated for a reference of 10 volts RMS across a 600-ohm resistive load. The frequency response remained flat within ± 1dB of the reference voltage from 50 to 10,000 Hz.

Harmonic distortion. The harmonic distortion of the booster amplifier was evaluated with a Distortion Analyzer (Hewlett Packard Model 333 A). The harmonic distortion generated by a 1,000-Hz tone was measured at two different levels. These levels were 1 volt RMS across a 600-ohm resistive load, and 10 volts RMS. The observed harmonic distortion values for both output voltages were less than 1 per cent.

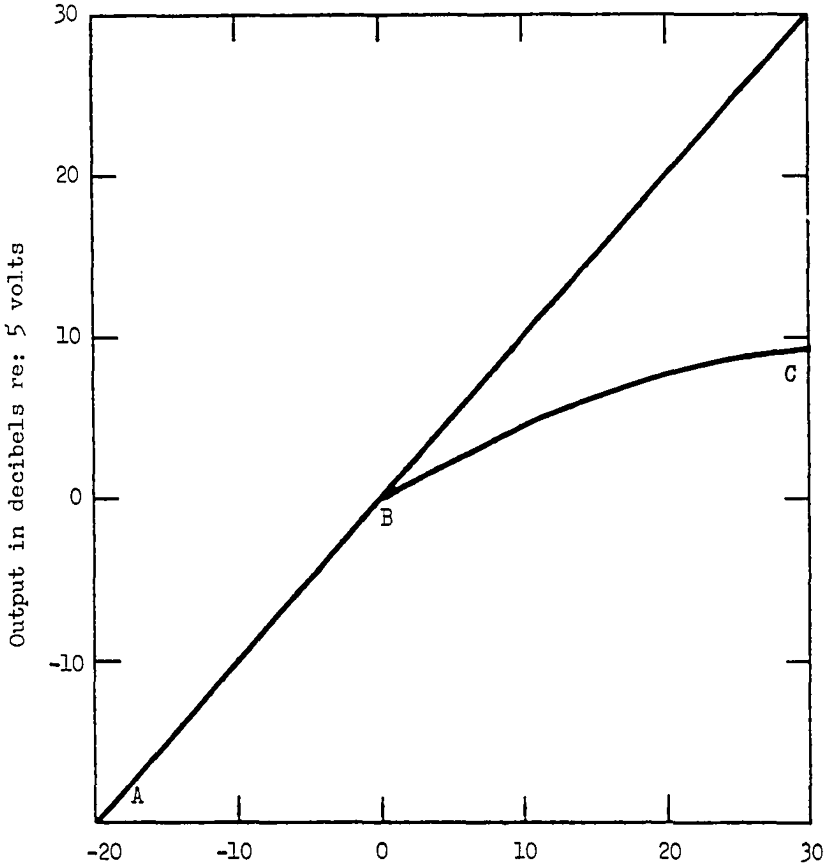
The Compressor Channel

The basic characteristics of the compressor amplifier (Altec 436 Compressor Amplifier) important for the present investigation were the frequency response, the characteristics of the input-output function, the amount of harmonic distortion, and the time constants. The evaluation of these characteristics is discussed in the following subsections.

Frequency response. The frequency response of the compressor amplifier was evaluated at three levels: 5dB below the threshold of compression, 10dB above the threshold of compression, and 20dB above the threshold of compression. A 1,000-Hz tone was applied from a signal generator to the compressor through an attenuator. Utilizing the output signal voltages generated by the 1,000-Hz tone as a point of reference, the output at other frequencies from 50 to 10,000 Hz was compared with the reference voltage. At all three levels the frequency response was flat within ± 1dB between 50 and 10,000 Hz.

Input-output function. The three major characteristics of the input-output function of the compressor amplifier are the threshold of compression, the range of the compression region, and the slope of the compression region or the compression ratio. A graph of the input-output function of the specific compressor amplifier used in this study appears in Figure 4.

Threshold of compression. The threshold of compression was measured according to the procedure set forth by Grimwood (22). A 1,000-Hz tone was applied to the compressor amplifier and the level of the signal generated at the output of the compressor was measured for both the uncompressed and compressed modes. The uncompressed mode was achieved by removing the compressor tube and disabling the compressor circuit. The level of the 1,000-Hz tone was increased by adjusting the gain control of the compressor until the level of the signal appearing at the compressor's output was one-half a decibel below the uncompressed



Input in decibels re: input necessary to achieve threshold of compression (5 volts)

Figure 4--A graph of the input-output function of the specific Altec 436 Compressor Amplifier used in the present experiment.

output level for the same input signal. Once this level was set, the level of the signal was measured at the output of the compressor. The threshold of compression was measured to be 5 volts. Point B in Figure 4 represents the threshold of compression for the experimental unit.

Range of the compression region. The range of the compression region was measured by establishing the threshold of compression and the saturation level. The measurement of the threshold of compression is discussed in the preceding section. In order to measure the unit's saturation level the output of the compressor was viewed on an oscilloscope. The level of the 1,000-Hz tone was increased until the waveform at the output of the compressor began to be clipped. The useful working range of the compressor was recorded as the difference between the level of the 1,000-Hz tone required to activate the compression circuit and the level required to drive the compressor into saturation. Distortion became apparent for this particular unit when the input signal exceeded that which drove the amplifier to the threshold of compression by approximately 30 dB. The resulting output range for the 30dB increase in the input signal was 9dB. This range of the compression region is represented by the line joining points B and C in Figure 4.

Slope of the compression region or the compression ratio. The slope of the compression region is defined as the output range divided by the input range. For this particular unit the output range of 9dB was divided by the input range of 30dB resulting in a value of 0.3. The slope of the compression region can also be expressed as a ratio of the range of the input signal above the threshold of compression to the range of the resultant output. The compression ratio for this unit was 30:9 or 3.3:1.

<u>Harmonic distortion</u>. The harmonic distortion of the 436 Compressor Amplifier was measured with a Distortion Analyzer (Hewlett Packard Model 333 A). The harmonic distortion generated by a 1,000-Hz tone was measured at three different levels: 5dB below the threshold of compression, 10dB above the threshold of compression, and 20dB above the threshold of compression. At all three levels the observed harmonic distortion value was less than 1 per cent.

<u>Time constants</u>. The 436 Compressor Amplifier is designed with an attack time of 50 msec and a release time which is variable from .33 to 1.33 seconds. These values were specified by the manufacturer at the conventional 63 per cent point of the time required for the unit to complete a change in gain. Grimwood (<u>22</u>) states that, "the measurement of action times therefore resolves itself into the measurement of the growth and decay times of the envelope of an audio-frequency signal."

To measure the attack time of a compressor amplifier, Grimwood recommends that a signal envelope of sufficient magnitude to activate the compression circuit should be applied to the amplifier. To achieve this condition, a 1,000-Hz tone which was 20dB greater than that required to reach the threshold of compression was applied to the compressor via a timing and gating apparatus which consisted of an Interval Timer (Grason Stadler Model 471) which controlled an electronic switch (Grason Stadler Model 829 C). The simplified block diagram in Figure 5 represents the equipment used for the measurement of the attack time. The interval timer was used to trigger the gating of the signal to the compressor and to synchronize the calibrated sweep of a storage scope (Tektronix Model 564 B). The time required for the compressor to achieve 63 per cent of its change in gain from the uncompressed signal

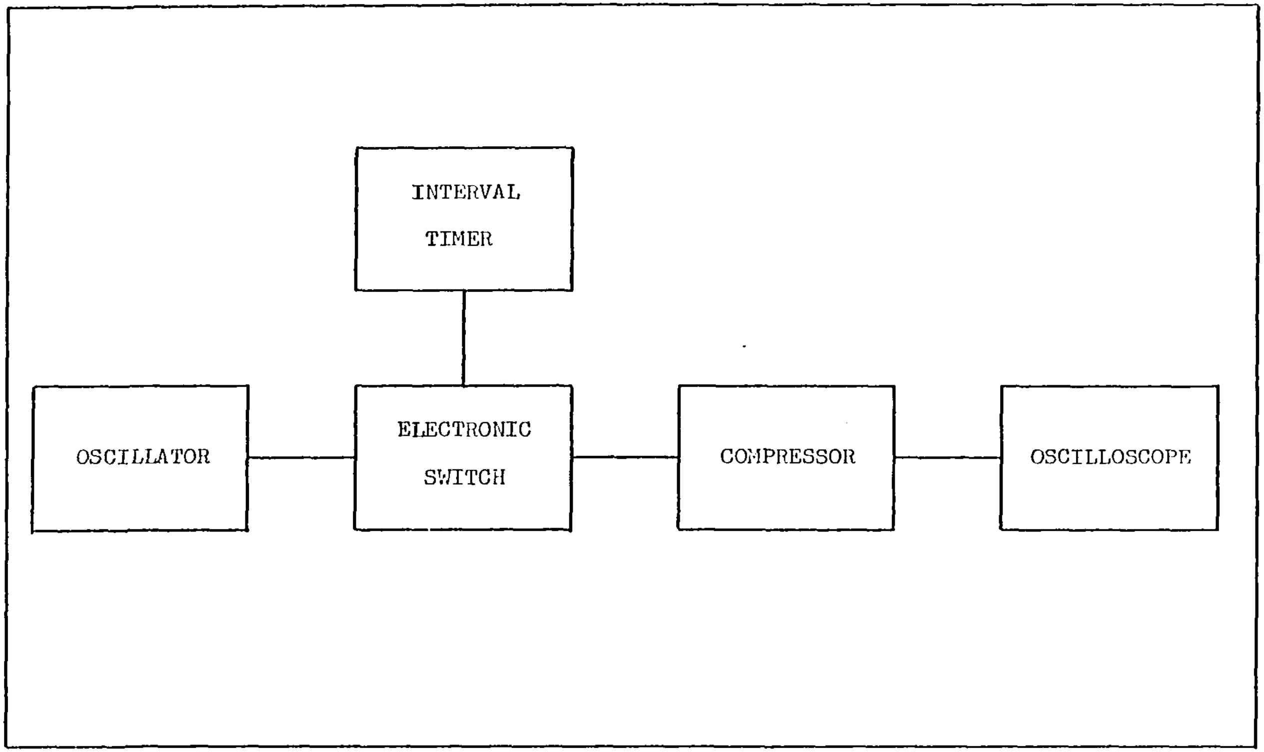


Figure 5--A simplified block diagram of the equipment used for the measurement of the attack time of the compressor amplifier.

level to steady-state compressed level was measured as the attack time. The attack time measured in this way was 50 msec.

To measure the release time of the compressor amplifier, Grimwood (22) advocates "switching a high audio-frequency input signal from a level which gives a chosen amount of compression to a lower level which gives an output below the threshold level." The simplified block diagram shown in Figure 6 represents the equipment used for the measurement of the release time.

The output of the signal generator was divided into two separate channels designated as SIG.-1 and SIG.-2 in the diagram. The level of Signal 1 was set to exceed the threshold of compression by 20 dB. This signal was applied to the compressor through a resistive mixer by means of the same timing and gating apparatus used in the measurement of the attack time. The duration of the signal was adjusted so that the signal envelope appeared only during the first half of the sweep of the storage scope. The level of Signal 2 was adjusted to a value which was approximately 5dB below the threshold of compression. This signal was routed through an attenuator and the mixer to the compressor amplifier. The release time was measured as the time required for the compressor to change its gain from the cessation of the larger signal to 63 per cent of the ultimate amplitude of the steady-state smaller signal. Although the shortest nominal release time of the compressor as specified by the manufacturer was .33 seconds, the experimental unit was capable of a release time of 300 msec. This release time was accepted as compatible with the requirements of the present study.

The Clipper Channel

The basic characteristics of the clipper-amplifier important

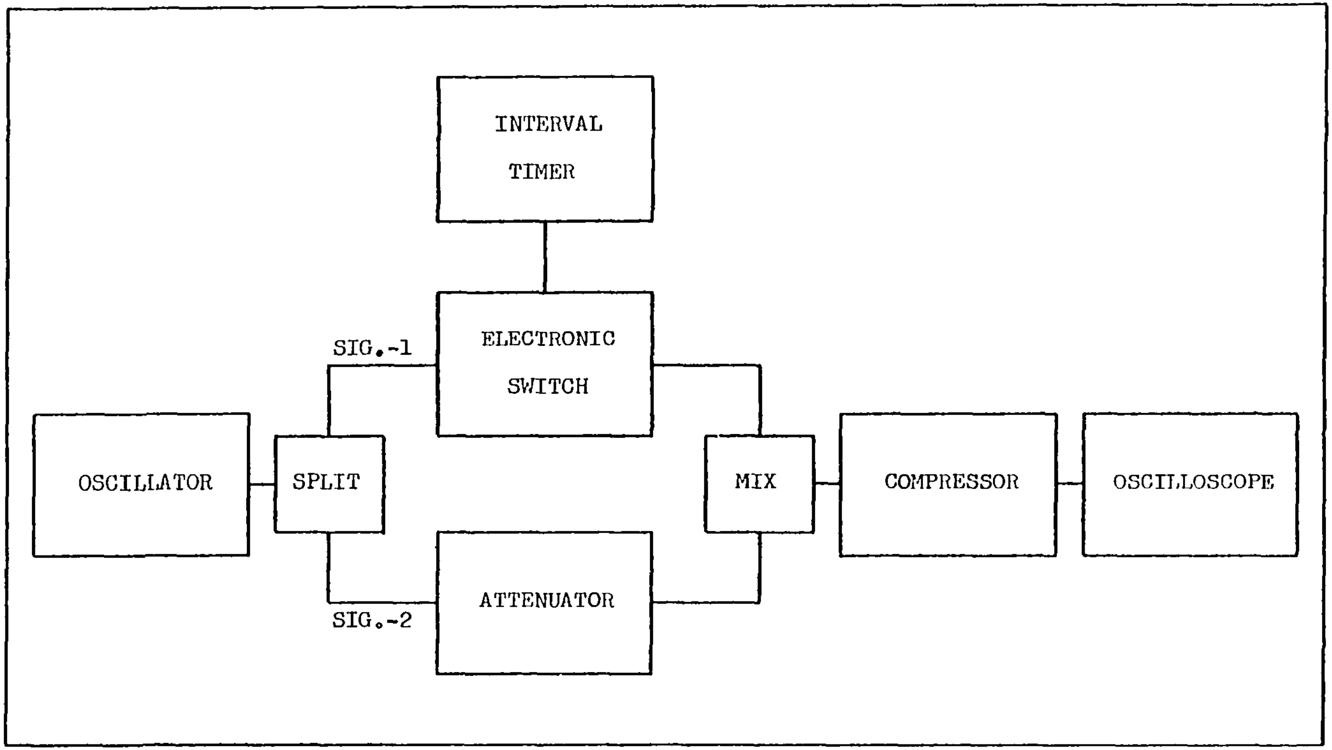


Figure 6--A simplified block diagram of the equipment used for the measurement of the release time of the compressor amplifier.

for the present investigation were its frequency response and the amount of harmonic distortion that it produced. The measurement of these characteristics is discussed in the following sub-sections.

Frequency response. The frequency response of the clipper was evaluated at a setting which was below the threshold of clipping. A 1,000-Hz signal from a generator was introduced to the input of a voltmeter (Ballantine 300) used as a clipper. The level of the signal was adjusted to achieve a value of 1 volt across a 15,000-ohm resistor terminating the output of the meter. The output voltages generated by signals ranging from 50-10,000 Hz were compared to this reference voltage. The frequency response was found to be flat within ± 1dB from 50-10,000 Hz.

Harmonic distortion. The harmonic distortion of the clipper was measured with a Distortion Analyzer (Hewlett Packard Model 333 A). The harmonic distortion generated by a 1,000-Hz tone was measured for two different input levels. The first level, 1 volt RMS across a 15,000-ohm resistor load, was selected to be below the threshold of clipping. The observed harmonic distortion value was less than 1 per cent. The second level was 20dB beyond the threshold of clipping. The observed harmonic distortion was 36 per cent.

Adjustment and Calibration of Equipment

The arrangement of the experimental apparatus adopted for the study permitted the simultaneous processing of speech and noise, combined at specified signal-to-noise ratios, through three separate channels. One channel provided non-limited speech, a second channel provided compressed speech and the third channel provided clipped speech.

Each of the three signals was then presented to one of three subjects at a level of 40dB above his speech reception threshold. In order to achieve the desired signal-to-noise ratios, amounts of compression and clipping, and presentation levels, the following adjustments were made to the equipment used for the playback of the experimental tapes.

Signal-to-Noise Ratios

The specific signal-to-noise ratios used in the experiment were selected to provide both favorable and unfavorable listening conditions. It was intended that under the most favorable condition the subjects would score close to 100 per cent, whereas for the most unfavorable condition they would score approximately 10 per cent. It was also considered desirable to explore the subjects' performance under at least two intermediate conditions. Pilot studies indicated that the use of -5, 0, and + 5dB signal-to-noise ratios and quiet would result in the desired levels of performance.

The experimental equipment was arranged so that the signal-to-noise ratios could be set by manipulation of attenuators 1 and 2 (refer to Figure 3). In order to maintain a constant amount of clipping and compression of the speech signal, the level of attenuator 1 was fixed and only attenuator 2 (the noise attenuator) was used to set the desired signal-to-noise ratios. Attenuator 1 was maintained at 5dB of attenuation. In order to achieve a +5dB signal-to-noise ratio the noise attenuator was adjusted to 0dB of attenuation. With these settings, the level of the speech signal was 5dB lower than the level of the noise signal. By placing 5dB of attenuation in attenuator 2 (the noise attenuator), the level of the noise was decreased 5dB and the resulting signal-to-noise ratio was 0dB. A further decrease in the level of the noise by introducing a total of 10dB of attenuation resulted in a +5dB signal-to-noise ratio. The quiet condition was obtained by using maximum attenuation (110dB) in attenuator 2. After adjustment, the signals were mixed and routed to the booster amplifier.

Setting the Amount of Clipping and Compression

It was arbitrarily decided to use 20dB of compression and clipping for the present investigation. Rather than using a speech signal with its constant fluctuation as the reference for setting and measureing the amounts of limiting, it was decided to use a 1,000-Hz sine wave. The level of the sinusoidal signal was adjusted to the same VU meter reading as that attained by the peaks of the speech signal. Since the electrical peaks of speech exceed the VU peaks obtained by the speech signal by 8 to 12dB, it was realized that the instantaneous peaks of the signal would be limited by more than 20dB. For measurement purposes, however, the level of the 1,000-Hz sine wave was set to exceed the threshold of the compressor and clipper by 20dB.

In order to measure the amounts of limiting for both the compressor and the clipper, it was necessary first to establish the threshold values of the experimental units. The level of a tone introduced from an oscillator was adjusted to a VU meter reading of -10. The output of attenuator 1 was channeled to the mixer and then to the booster amplifier. The output of the booster amplifier was adjusted by means of its gain control to a VU meter reading of +10. The amplified signal was then routed to attenuator 3 which was set for 20dB of attenuation. The output of attenuator 3 passed to the three signal-processing channels.

The gain control of the compressor was adjusted so that the level of the 1,000-Hz tone was just below the threshold of the compressor. Once this level was set, removal of the 20dB of attenuation in attenuator 3 resulted in a signal level which drove the compressor into compression by 20dB.

The threshold of the clipper was established by monitoring the output of the unit visually on an oscilloscope. The potentiometer preceding the clipper, labeled POT in the diagram, was adjusted so that the output signal appearing on the scope was just barely clipped. Since the positive peaks of the signal were clipped slightly more than the negative peaks, the decibel difference between these levels was computed and the gain control was set for a median value. Upon removal of the 20dB of attenuation in attenuator 3, the signal was clipped by 20dB. The resulting trace resembled a square-wave when viewed on the oscilloscope.

Adjusting the Presentation Level

It was established in a preliminary study that a presentation level of at least 40dB SL was required for subjects to be able to identify at least a few of the test words under the most adverse listening condition. It was decided to adopt this level for the experiment.

Attenuators 4, 5, and 6 (refer to Figure 3) were used to set the level of the signal delivered to the earphones. In order to obtain a roughly equivalent reading at the earphones for a given attenuator setting, it was necessary to adjust the output level of each channel prior to the attenuators. It was decided to adjust the output levels of the compressed and clipped channels to that of the non-limited channel because the signal level was lowest in that channel. This adjustment was made for the clipper by manipulation of the gain control of the impedance converter, labeled AMP.2 in the diagram. For the compressor this adjustment was made by manipulation of the resistive T pad labeled T-PAD in the diagram. The approximate equivalence of these three signal levels was confirmed electrically at the output of attenuators 4, 5, and 6, and acoustically at the output of the earphones. The electrical measurement was obtained with a voltmeter (Ballantine 300) bridged across the output of each attenuator and the acoustical measurement was obtained at the output of the earphone with an artificial ear (Western Electric 640-AA Condensor Microphone and Western Electric Acoustic Laboratory, Type 100 D/E Condensor Microphone Complement), using the same Ballantine meter as a readout device. The frequency response measurement performed on each earphone confirmed that the earphones which were used performed as a matched set within ±1dB over the frequency range of interest.

Instructions and Test Procedures

The test session was divided into four portions. The first portion consisted of the instructions given to the subjects and the familiarization of the subjects with the test materials. The second, third, and fourth portions consisted of the tests conducted under the three main experimental conditions. A test session was completed in approximately one hour and forty-five minutes.

Three subjects were seated in the examinee's room during each experimental session. Each subject was provided with his own desk, response sheets, and a headset. A VU Meter (Weston 802), used to cue the response of the subjects, was located in view of all three individuals. The VU Meter enabled the subjects to keep track of each test time, particularly during the most difficult listening condition.

Once the subjects were seated, the examiner repeated the following instructions:

> This test consists of two listening tasks. The first listening task involves the identification of the words listed on this paper. After you have read the list you will be fitted with earphones. You will hear the speech signal only in your right ear. A male talker will introduce the word list and then proceed to present the words. After each word you will have a five-second interval in which to repeat the word. I will be able to hear your response through the overhead microphone. The first word will be fairly loud but the following words will gradually become softer. Repeat each word that you possibly can. Only one of you will hear the words at a time. (name), you will be first, ________, will be second, and ________, you will be last. Are there any questions?
>
> Before we proceed with this portion of the test, I am going to describe the second listening task because once the earphones are in place, they will not be removed until both portions of the test are completed.
>
> The second listening task involves the identification of the words listed on this paper. There will be fifty words in each list. You will notice that your response sheets have a blank for each of the fifty words. Again, you will hear a male talker introduce the word lists. He will then ask, "Are you ready?" The talker will introduce each word with the phrase, "Say the word...." You will have a four and one-half second interval in which to record your response. During some of the word lists, you will hear noise which may or may not interfere with your identification of the words. To assist you in tracking the occurrence of each word, you will notice that the needle on this meter will deflect during each carrier phrase and word. If you cannot identify the word, draw a line through the appropriate blank. Do not become concerned if you cannot identify many of the words. This is to be expected under the noisiest conditions. During this part of the test, all three of you will receive the words at the same time. Are there any questions?

Following the instructions, the earphones were placed on each subject. The recording of spondee words was reproduced through the active earphone of one subject at a time. Attenuators 4, 5, and 6 were used to establish a speech reception threshold for each subject. Testing was begun with 60dB of attenuation inserted in the attenuator being used to establish threshold and 110dB of attenuation inserted in the attenuator of the other two channels. The presentation level was reduced in 10dB steps until the subject could not repeat any of the words. The level was then increased by 8dB and each presentation, thereafter, was reduced in 2dB steps. The speech reception threshold was recorded as the lowest level at which the subject correctly identified two out of four words. This procedure was followed for each of the three subjects.

Once the speech reception threshold for each subject was established, attenuators 4, 5, and 6 were adjusted to provide 40dB less attenuation. This established a presentation level of 40dB above each subject's speech reception threshold.

In order to reduce the effects of any systematic biases on the data, the order of presentation of the experimental conditions was counterbalanced. Six of the eighteen subjects tested in this study listened to the non-limited condition first, six listened to the compressed condition first and six listened to the clipped condition first. Six individuals heard List II of N.U. Auditory Test No. 6 first, six heard List III first, and six heard List IV first. The order of presentation of signal-to-noise ratios and quiet, however, was the same for all subjects: -5, 0, +5dB signal-to-noise ratios and quiet. The subjects proceeded in this order from the most difficult listening condition to the easiest. This order of presentation was adopted to reduce the possibility of biasing the speech discrimination scores according to the reasoning used by Tillman and Carhart (<u>62</u>) during the development of N.U. Auditory Test

No. 4.

The appropriate tape was placed on the recorder as soon as test conditions were selected according to the counterbalancing schedule. The subjects listened to the appropriate material at signal-to-noise ratios of -5, 0, and +5dB and in quiet, in that order. After the presentation of the four word lists, the subjects were provided with a short rest period. Following this the subjects rotated seating positions. Subject 1 moved to position 2, subject 2 moved to position 3, and subject 3 moved to position 1. This procedure was repeated after each listening session until each subject had listened to each of the experimental conditions.

Calibration Recheck

The frequency response and harmonic distortion of the booster amplifier, the compressor, and the clipper were remeasured immediately following the collection of the experimental data. All the measurements were in agreement with those obtained prior to the collection of the data except that the frequency response of the compressor was found at this time to be within ±2dB of the reference voltage. This measurement reflected a slight change from the original frequency response obtained prior to the collection of the data. The harmonic distortion values were observed to be 1.6 per cent below the threshold of compression and approximately 5 per cent above the threshold of compression.

These distortion values were slightly poorer than those obtained prior to the collection of the data. Apparently, a slight change in the characteristics of the compressor, undetectable by routine calibration check procedures carried out before each listening session, had occurred at some time between the initiation and the completion of testing. The decision was made, nevertheless, to accept the data as satisfactory for the purpose of the experiment because the results closely resembled those obtained in an earlier pilot study. This study used essentailly the same procedures and the same number of subjects as the main study, and the outcomes of statistical treatment of the data were the same as those obtained in the main study.

Data Analysis Procedures

Three speech reception thresholds and twelve speech discrimination scores were obtained for each subject. The speech reception thresholds were expressed in terms of the sound pressure levels required to establish a threshold response. Each speech discrimination score was expressed as a percentage of the words correctly identified out of each list of fifty words.

The speech reception thresholds for all subjects over each experimental condition were tabulated and the means, medians, ranges, and standard deviations of the means were calculated. The threshold values were also treated statistically in an analysis of variance based on a completely randomized block design.

The speech discrimination scores were treated statistically in an analysis of variance procedure using a 3 x 4 x 18 factorial design with repeated measurements on each factor. The results of the study and a discussion of these results are presented in the following chapter.

CHAPTER IV

RESULTS, DISCUSSION, AND CONCLUSIONS

Introduction

The purpose of the present investigation was to compare the intelligibility of non-limited, compressed, and clipped quiet and noisy speech signals. In order to achieve this purpose, an apparatus was designed which permitted the combining of speech and noise in any desired ratio. The combined signal and noise were processed through three separate channels of the apparatus. One of these channels provided unaltered or non-limited reproduction, one provided signal compression, and one provided peak clipping. Each channel was terminated with an attenuator used to adjust the level of the signal presented to a subject's earphone.

Eighteen normal-hearing young adults served as subjects for the experiment. Speech reception thresholds for non-limited, compressed, and clipped W-2 Spondee Words were obtained for each subject in quiet. The thresholds obtained under each condition served as the reference values for the presentation of speech discrimination tests under the same condition. These tests were presented at a level of 40dB above each subject's speech reception threshold. The test materials used for the speech discrimination tests consisted of taped copies of Sommerville's recordings of Lists II, III, and IV of the N.U. Auditory Test No. 6 word lists. Speech discrimination scores were obtained for each subject at four different signal-to-noise ratios under three conditions of amplification for a total of twelve scores per subject.

The following sections of this chapter contain a description of the speech reception thresholds and speech discrimination scores obtained in the course of the study. The descriptive statistics presented in this chapter were derived from data processed at the Computer Center of the University of Oklahoma Medical Center.

Speech Reception Thresholds

Three speech reception thresholds were obtained for each subject, one for non-limited speech, one for compressed speech, and one for clipped speech. All three speech reception thresholds were obtained under the quiet condition, that is, with no noise mixed with the speech.

The individual speech reception thresholds obtained for each subject under every experimental condition are recorded in Table 9, located in Appendix D. The means, medians, ranges, and standard deviations of the speech reception thresholds averaged over all eighteen subjects appear in Table 1. These values are recorded in terms of the approximate sound pressure levels required to establish a threshold response.

The mean speech reception threshold for non-limited speech was found to be 19.4 dB SPL, that for compressed speech was 22.8dB SPL and that for clipped speech was 21.2dB SPL. Based on these mean values, it appears that non-limited speech required the least sound pressure level to reach threshold, clipped speech required the next lowest sound pressure level, and compressed speech required the greatest sound pressure level. The largest difference occurred between the non-limited and

TABLE 1

THE MEANS, MEDIANS, RANGES AND STANDARD DEVIATIONS OF THE SPEECH RECEPTION THRESHOLDS EXPRESSED IN DECIBELS RE: 0.0002 MICROBAR AND AVERAGED OVER EIGHTEEN SUBJECTS FOR NON-LIMITED, COMPRESSED, AND CLIPPED SPEECH

Condition	Mean	Median	Range	Standard Deviation
Non-Limited	19.4	19.0	15-23	2.12
Compressed	22.8	23.0	17-27	3.59
Clipped	21.2	21.0	13-29	5.82

compressed conditions of amplification. This difference was 3.4dB.

The median speech reception threshold obtained for non-limited speech was 19dB SPL. The corresponding values for the other two experimental conditions were 21dB SPL for clipped speech and 23dB SPL for compressed speech. Again, the largest difference occurred between the non-limited and compressed conditions of amplification. This difference was 4dB.

The ranges of the individual speech reception thresholds obtained for non-limited and compressed speech were relatively small. The range was 8dB for non-limited speech and 10dB for compressed speech. The standard deviations for both of these conditions were correspondingly small, 2.1dB for non-limited speech and 3.6dB for compressed speech. The 16dB range of the speech reception thresholds obtained for clipped speech was somewhat larger than that observed for the other two conditions. This increased range was reflected in a standard deviation of 5.8dB.

Statistical Treatment and Comparison with Other Studies

In order to test the significance of the differences observed among the mean speech reception thresholds obtained under the three conditions of amplification the data were treated in an analysis of variance (AOV) with repeated measures for each condition of amplification. Statistical comparisons were made among the treatments using an analysis similar to that described by Steel and Torrie (59) for a completely randomized block design.

The results of the AOV appear in Table 2. The analysis shows

TABLE 2

RESULTS OF THE ANALYSIS OF VARIANCE FOR THE SPEECH RECEPTION THRESHOLDS OBTAINED WITH NON-LIMITED, COMPRESSED, AND CLIPPED SPEECH FOR EIGHTEEN SUBJECTS

Source of Variation	df	SS	MS	F
Treatments (Types of Amplification)	2	100.1	50.1	6.1 *
Blocks (Subjects)	17	493.1	29.0	3.6 *
Error	34	277.5	8.2	

* Significant at the .01 level

that a significant difference exists among types of amplification ($p < .01$). A significant difference is also observed among subects ($p < .01$).

Type-of-amplification effect. The results of the analysis of variance indicate that a statistically significant difference exists among the types of amplification used in the conduct of the experiment.

Although the AOV indicates that at least one of the three treatments differs significantly from the others (first line of AOV significant at .01 level) the analysis does not explore the relationship among the three means. Duncan's New Multiple Range Test was selected as the a' posteriori testing procedure for individual comparisons of pairs of means. The results of the procedure are summarized in Table 3. In the table Condi-

TABLE 3

SUMMARY OF COMPARISONS BETWEEN THE MEAN SPEECH RECEPTION THRESHOLDS OBTAINED UNDER THE NON-LIMITED, COMPRESSED, AND CLIPPED CONDITIONS OF AMPLIFICATION USING DUNCAN'S NEW MULTIPLE RANGE TEST

Conditions (1-NL, 2-CP, 3-CL)	Decibel Differences Between Means	Value Required for Significance at .01	Results
1-2	3.4	1.94	1 2 3
1-3	1.8	2.05	
2-3	1.6	2.05	

tion 1 corresponds to non-limited speech, designated as NL in the column heading. Condition 2 corresponds to compressed speech and is designated as CP while Condition 3 represents clipped speech, designated as CL. The lines drawn in the results column connect those conditions for which no difference was observed.

The absolute difference between the mean speech reception thresholds for non-limited and compressed speech (1-2) is 3.4dB. The corresponding differences between the non-limited and clipped speech reception thresholds (1-3) and the compressed and clipped speech reception thresholds (2-3) are 1.8dB and 1.6dB, respectively. The results of the statistical procedure indicate that the differences between the non-limited and clipped speech reception thresholds, and the compressed and clipped speech reception thresholds fail to achieve significance at the .01 level. The difference between the non-limited and compressed speech reception thresholds, however, was found to be significant at this level.

The results were compared to those obtained by Caraway (8) in her investigation of the effects of dynamic range reduction with a compressor amplifier on speech intelligibility. Caraway's means, medians, and standard deviations for non-limited and compressed speech are presented in Table 4. The results of the present experiment for non-limit-

TABLE 4

A COMPARISON OF THE MEANS, MEDIANS, AND STANDARD DEVIATIONS OF THE SPEECH RECEPTION THRESHOLDS EXPRESSED IN dB RE: 0.0002 MICROBAR OBTAINED IN CARAWAY'S STUDY AND THE PRESENT INVESTIGATION UNDER CONDITIONS OF NON-LIMITED AND COMPRESSED AMPLIFICATION

Statistic	Non-Limited		Compressed	
	Caraway	Present Study	Caraway	Present Study
Mean	20.0	19.4	18.7	22.8
Median	20.0	19.0	20.0	23.0
Standard Deviation	2.1	2.1	3.2	3.6

ed and compressed speech also appear in the table to facilitate comparison.

The mean speech reception threshold for non-limited speech in

Caraway's study was 20.0dB SPL. The mean speech reception threshold for non-limited speech in the present experiment was 19.4dB SPL. These values are remarkably similar, differing by only 0.6dB. The mean speech reception threshold for compressed speech in Caraway's study was 18.7dB SPL, while the mean speech reception threshold for compressed speech in the present investigation was 22.8dB SPL. These values differ by 4.1dB.

Comparison of the mean speech reception thresholds obtained by Caraway using non-limited and compressed speech shows that, on the average, a lower sound pressure level was required for the establishment of a threshold response for compressed speech than for non-limited speech. A comparison of the mean speech reception thresholds obtained for non-limited and compressed speech in the present experiment indicates that a lower sound pressure level was required to elicit a threshold response for non-limited speech. In other words, the direction of the difference between non-limited and compressed speech was opposite in the two studies.

The median speech reception thresholds for non-limited speech obtained for both studies differ by only 1dB and the median speech reception thresholds obtained for compressed speech for both studies differ by 3dB. The standard deviations of the mean speech reception thresholds for both non-limited and compressed speech in both studies are in good agreement.

The differences observed between the values obtained for compressed speech in the two studies may possibly be accounted for by the fact that Caraway used different subjects, test materials, apparatus and experimental procedures. Nevertheless, the threshold values obtained in the present experiment for both compressed and non-limited speech appear to be in fairly good agreement with those obtained by Caraway.

Subject effect. The results of the analysis of variance in Table 2 show that a significant difference existed among subjects (P .01). Although it was not elected to explore these differences statistically, it was considered important to report a possible source of variation which could be observed upon inspection of the data. Reference to the responses of the individual subjects reveals that the speech reception thresholds varied under the clipped condition over a relatively wide range compared to the narrower range of speech reception thresholds obtained for non-limited and compressed speech. The dispersion of the subjects' responses observed for clipped speech was reflected in a standard deviation of 5.8dB.

Variability in subject response for clipped speech has been reported previously. Licklider and Pollack (35) attributed this variability to the skills manifested by individuals in their attempt to decode distorted information and the diligence applied to the task. There was no reason to suppose that the subjects in the present investigation would perform identically. Therefore this particular known source of variation was partitioned out in the AOV so that it would not appear as treatment differences.

Discussion

The outcome of the statistical treatment of the speech reception thresholds obtained for the three different experimental conditions indicated that the thresholds obtained under the compressed condition were significantly different from those obtained under either of the two other conditions. While this is no doubt correct for the present data, it would seem imprudent to attempt to generalize to a wider population for at least two reasons.

First, the speech reception thresholds obtained with the clipped signal may have been in error by as much a 1dB, because an average responding voltmeter was used to equate the levels in the three signal channels. Such a meter is susceptible to an error of this magnitude for signals which are rich in harmonic content, such as those altered by clipping. The use of a true RMS voltmeter would have eliminated this error but such an instrument was not available at the time the experiment was conducted.

Second, it is possible that the slight change in frequency response of the compressor previously mentioned in Chapter III may have affected the level of the compressed signal. This seems plausible since the difference observed between the speech reception thresholds for compressed and non-limited speech in the present experiment was not observed in a pilot study in which thresholds for both non-limited and clipped speech were obtained under similar conditions using the same compressor amplifier. The absolute difference between compressed and non-limited speech was only .8dB in the pilot study, and the direction of the difference was the same as in the main experiment. Caraway observed a small (1.3dB) mean difference in the opposite direction.

Since the speech reception thresholds served simply as reference values for setting the presentation levels of the word lists, the problem of interpretation is not considered to be of great importance, particularly since the absolute differences observed among all three conditions are small and probably clinically insignificant.

Speech Discrimination Scores

Twelve different speech discrimination scores were generated by each subject during the course of the experiment. Four of these twelve scores represented the subject's performance for non-limited speech at signal-to-noise ratios of -5dB, 0dB, +5dB and in quiet. Four scores represented his performance for compressed speech and four scores represented his performance for clipped speech under these same signal-to noise conditions.

The speech discrimination scores obtained for every subject under each of the twelve experimental conditions appear in Table 10 of Appendix E. Each score represents the percentage of words correctly identified out of a list of fifty words. The mean speech discrimination scores obtained under the three experimental conditions at each of four different signal-to-noise ratios appear in Table 5.

TABLE 5

THE MEAN SPEECH DISCRIMINATION SCORES EXPRESSED IN PER CENT OBTAINED WITH NON-LIMITED, COMPRESSED, AND CLIPPED SPEECH AT SIGNAL-TO-NOISE RATIOS OF -5dB, 0dB, +5dB, AND QUIET

	Signal-to-Noise Relationship			
Type of Amplification	-5dB	0dB	+5dB	Quiet
Non-Limited	8.7	49.8	81.5	98.8
Compressed	7.5	45.4	77.4	98.3
Clipped	2.4	14.9	30.4	70.4

The mean speech discrimination scores obtained under each of the experimental conditions are plotted as a function of signal-to-noise ratio in the graph making up Figure 7. It appears, by inspection, that compressed and non-limited speech yielded similar results, but that the scores obtained under the clipped condition are inferior at all four signal-to-noise ratios. The mean speech discrimination scores for non-limited speech are 8.7 per cent for the -5dB signal-to-noise ratio, 49.8 per cent for the 0dB signal-to-noise ratio, 81.5 per cent for the +5dB signal-to-noise ratio, and 98.8 per cent for the quiet condition. Following the same order of signal-to-noise conditions, the mean speech discrimination scores for compressed speech are 7.5 per cent, 45.4 per cent, 77.4 per cent, and 98.3 per cent, and for clipped speech are 2.4 per cent, 14.9 per cent, 30.4 per cent and 70.4 per cent. The largest difference exhibited between the non-limited and compressed-speech discrimination scores occurs at the 0dB signal-to-noise ratio. This difference is 4.4 per cent. The largest difference between the speech discrimination scores obtained for non-limited and clipped speech occurs at the +5dB signal-to-noise ratio. This difference is 51.1 per cent. The largest difference between the speech discrimination scores obtained for compressed and clipped speech occurs also at the +5dB signal-to-noise ratio. This difference is 47 per cent.

Statistical Treatment and Comparison with Other Studies

In order to test the significance of these and other differences, the experimental data were treated in a 3 x 4 x 18 factorial analysis of variance. For statistical purposes the repeated measure aspect of the design (each subject measured under all twelve experimental conditions) was accounted for by considering subjects as a random factor

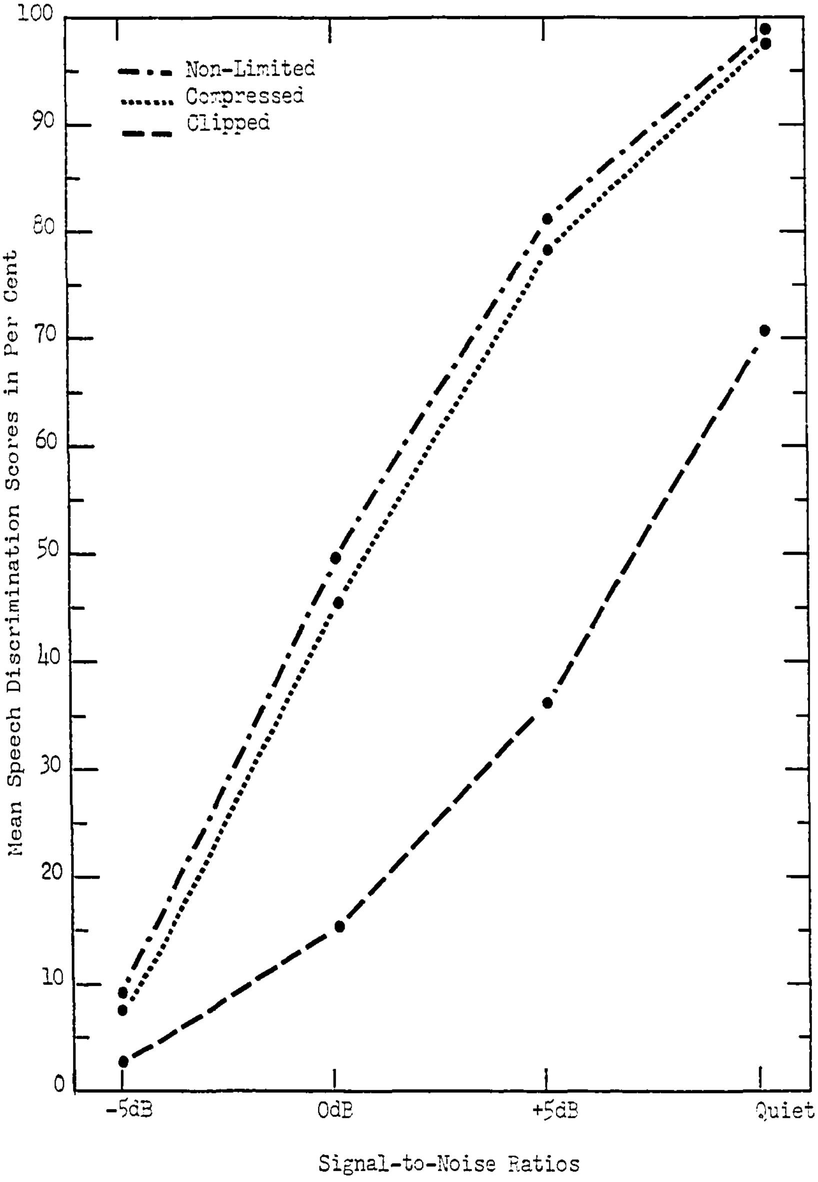


Figure 7--A plot of the mean speech discrimination scores obtained for non-limited, compressed, and clipped speech at signal-to-noise ratios of -5dB, OdB, +5dB, and quiet.

appearing at eighteen levels and testing and interpretation made in the context of a mixed model analysis of variance. Selected statistical comparisons were made among the treatments by an analysis of variance (AOV) similar to that described by Winer (64) for a p x q x r factorial experiment in which treatment totals are used in computing sums of squares for testing hypotheses concerning main effects and interactions.

The results of the AOV performed for the treatment means for each experimental condition are presented in Table 6.

TABLE 6

SUMMARY OF THE ANALYSIS OF VARIANCE MADE FOR SELECTED COMPARISONS AMONG NON-LIMITED, COMPRESSED AND CLIPPED SPEECH DISCRIMINATION SCORES OBTAINED UNDER SIGNAL-TO-NOISE RATIOS OF -5dB, 0dB, +5dB AND QUIET FOR EIGHTEEN SUBJECTS

Source of Variation	df	SS	MS	F
Signal-to-Noise	3	204,984.0	68,327.1	2,285.8 *
Type of Amplification	2	40,284.0	20,142.0	673.6 *
Noise Levels x Amplification	6	11,645.0	1,940.9	65.0 *
Subjects	17	3,150.0	185.3	6.2 *
Subjects x Noise	51	1,548.8	30.4	1.0NS
Subjects x Amplification	34	1,208.7	85.6	1.2NS

* Significant at the .01 level

The results of the AOV show that the effect of signal-to-noise ratio on the intelligibility of speech is significant at the .01 level. The results also show that the effect attributable to type of amplification and the effect attributable to subjects are also significant at the .01 level. Both the subjects x noise and the subjects x amplification conditions failed to achieve significance at the .01 level. Each of the known sources of variation will be discussed separately in the following sections.

Signal-to-noise effect. The results of the AOV in Table 6 show that the effect of signal-to-noise ratio on the intelligibility of speech is significant well beyond the .01 level. This result was, of course, anticipated and is in agreement with the findings of numerous other investigators. The discussion of the effect of signal-to-noise ratio will be divided into two parts in the following sections. The results of the experiment obtained in quiet will be discussed first, followed by a discussion of the results obtained in noise.

Speech discrimination scores obtained in quiet. Sommerville (57) in recording the N.U. Auditory Test No. 6 speech materials obtained articulation-gain functions for 32 subjects in quiet. These lists were presented to the subjects at sensation levels of -4, 0, 4, 8, 12, 16, 20, 24, 28, and 32dB. Scores of 96 to 98 per cent were obtained when the presentation level was 32dB above the subject's speech reception threshold. It is evident in comparing Sommerville's data with that obtained in the present investigation for non-limited speech, that the subjects serving in the present study performed appropriately for this specific set of materials. The mean speech discrimination score of 98.8 per cent obtained in this investigation was slightly better than that obtained by Sommerville. This difference may be accounted for by the somewhat higher presentation level used in this investigation. Observation of the mean speech discrimination scores for compressed speech suggests that signal compression did not alter the subjects' performance on the speech discrimination task under quiet listening conditions. For clipped speech, however, intelligibility was decidedly inferior.

Speech discrimination scores obtained in noise. Miller (43) obtained discrimination scores for non-limited speech in noise. In order to facilitate comparison of Miller's data to the present data, it was necessary to extrapolate values corresponding to the signal-to-noise ratios used in the present study from his graph of intelligibility scores plotted as a function of signal-to-noise ratio. These scores were compared to those obtained in the present investigation for signal-to-noise ratios of -5dB, 0dB, and +5dB. Table 7 provides a comparison

TABLE 7

A COMPARISON OF THE SPEECH DISCRIMINATION SCORES EXPRESSED IN PER CENT AND ROUNDED TO THE NEAREST WHOLE NUMBER OBTAINED BY MILLER FOR NON-LIMITED SPEECH AT -5dB, 0dB, AND +5dB SIGNAL-TO-NOISE RATIOS WITH THOSE OBTAINED IN THE PRESENT INVESTIGATION FOR NON-LIMITED, COMPRESSED AND CLIPPED SPEECH UNDER SIMILAR CONDITIONS

Data	-5dB S/N	0dB S/N	+5dB S/N
Miller's Study	10	50	82
Present Data Non-Limited	9	50	82
Present Data Compressed	8	45	77
Present Data Clipped	2	15	30

of the mean speech discrimination scores obtained in the present study for *non-limited, compressed, and clipped* speech for three signal-to-noise ratios to those obtained by Miller for non-limited speech under similar conditions.

It is apparent in comparing Miller's data to those obtained under the non-limited condition in the present study that the results are virtually identical. This is not entirely surprising since Miller used similar noise and similar speech materials. The largest difference, only 1 per cent, occurs at -5dB signal-to-noise ratio. It is clear from both studies that as the level of the noise approaches and exceeds the level of the speech signal the intelligibility of speech deteriorated rapidly.

Type-of-amplification effect. The results of the AOV in Table 6 show that the effect attributable to type of amplification is significant at the .01 level. Inspection of Figure 7 shows that the speech discrimination scores obtained for non-limited and compressed speech are similar, but that the scores obtained for clipped speech are inferior to those obtained for the other two conditions of amplification.

The New Duncan's Multiple Range Test was used to test the significance of the differences observed among experimental conditions. The results of the test are summarized in Table 8. Condition 1 in the table corresponds to non-limited speech (designated as NL). Condition 2 corresponds to compressed speech (designated as CP), and Condition 3 represents clipped speech (designated as CL). The lines in the summary of results column in Table 8 connect those conditions for which no significant difference was observed.

The difference between means for the pairing of non-limited

TABLE 8

SUMMARY OF COMPARISONS BETWEEN CONDITIONS OF NON-LIMITED, COMPRESSED, AND CLIPPED SPEECH EXPRESSED IN PER CENT AND AVERAGED OVER SIGNAL-TO-NOISE CONDITIONS

Conditions (1-NL, 2-CP, 3-CL)	Difference Between Over-All Means	Values Required for Significance at .01	Summary of Results
1-2	2.5	2.80	1 2 3
1-3	30.1	2.95	____
2-3	27.5	2.95	

and compressed speech (1-2) is 2.5 per cent. The difference between means for the pairing of non-limited and clipped speech (1-3) is 30.1 per cent. The difference between means for the pairing of compressed and clipped speech (2-3) is 27.5 per cent. The results indicate that the difference between the pairs of non-limited and compressed speech is not significant at the .01 level. The differences between the pairs of non-limited and clipped speech, and of compressed and clipped speech were found to be statistically significant at the .01 level.

Caraway obtained speech discrimination scores for normal-hearing subjects for both non-limited and compressed speech. She established articulation-gain functions by presenting speech discrimination tests at sensation levels of 0, 8, 16, and 24dB. For the purpose of comparing the present data with that obtained by Caraway. The results recorded at the highest presentation level that she used were selected. For non-limited speech in quiet, Caraway obtained a mean speech discrimination score of 98.0 per cent as compared to a mean speech discrimination score obtained in the present investigation at 40dB SL of 98.8 per cent. Caraway obtained a mean speech discrimination score of 98.5 per cent for compressed speech in quiet as compared to a score of 98.5 per cent for the present investigation. These scores are virtually identical and suggest that compressed speech is as intelligible as non-limited speech under favorable listening conditions.

It is much more difficult to compare the speech discrimination scores obtained for clipped speech to those obtained in other investigations since many variables were found to influence intelligibility. The range of reported speech discrimination scores varied from 50 to 90 per cent, depending upon subject sophistication, difficulty of materials, size of test sample, frequency response used prior to clipping, and amount of clipping. All that can be said of the present data is that the mean score of 70.4 per cent obtained for clipped speech lies within this range and that this mean score is inferior to that obtained for non-limited and compressed speech.

The outcome is consonant with the earlier findings of Davis, Stevens, and Nichols (15), who reported in 1947 that the average speech discrimination scores obtained for three normal-hearing subjects and six hard-of-hearing subjects were higher for 30dB of compression than for 30dB of clipping.

Signal-to-noise x amplification interaction. The results of the AOV in Table 6 show that the noise level by amplification interaction is significant at the .01 level. The reason for this outcome is clarified upon inspection of the graph in Figure 7. The graph shows that while the curves representing the performance of the subjects under the non-limited and compressed conditions are very similar, that representing their performance under the clipped condition differs substantially from the other two conditions. The curve representing the subjects' performance for clipped speech is not simply displaced (the main effect), but it is also considerably bowed with respect to the other two curves (the interaction).

The similarity in the slope of the curves across signal-to-noise ratios for non-limited and compressed speech indicates that noise is no more detrimental to the intelligibility of compressed speech than it is for non-limited speech. For clipped speech, however, this trend does not hold true. The absolute difference in quiet observed between non-limited and clipped speech is 28.4 per cent. Upon the introduction of noise in the +5dB and 0dB signal-to-noise conditions, the absolute difference is increased to 51.1 per cent and 34.9 per cent, respectively. This increase in the absolute difference indicated that the addition of the distortion of noise to the already distorted clippped signal results in an even more rapid deterioration in the intelligibility of speech. This occurrence is not surprising in view of the results of previous studies which indicate that the accumulative effect of combining distortion from two different sources can be additive and possible multiplicative (23).

This interpretation does not appear to hold, however, at the -5dB signal-to-noise ratio where only slight differences are observed among results obtained under the three different conditions of amplification. Nevertheless, the intelligibility of clipped speech at this ratio was still inferior to that observed under the other two conditions.

Subject effect and interaction. The results of the AOV in Table 5 show that the subject effect is significant at the .01 level.

The partitioning out of known sources of variation from the error term, the sizeable number of subjects and the large number of experimental conditions all contributed to the sensitivity of the test used in the analysis of the data. It was not surprising that with this increased precision the test was sensitive to intersubject differences. These differences appeared to be distributed throughout the data and could not be accounted for in the subjects x noise interaction or the subjects x type of amplification interaction.

Conclusions and Discussion

Contemplation of the data generated by the study and the results of the statistical procedures applied to these data leads to the formulation of three primary conclusions. It may be that while these conclusions apply strictly to the particular circumstances of the present study, they may serve as useful guidelines in assessing the effects of signal limiting on intelligibility in other applications until more definitive information becomes available.

The first conclusion drawn from the study is that compressed speech is equally as intelligibile as non-limited speech over a broad range of signal-to-noise ratios when the signal and noise are combined before entering the communication systems. This conclusion is supported by inspection of the data, which shows similar results for both conditions. Moreover, the outcome of the statistical analysis shows that the small difference observed between the two conditions is not statistically significant.

If it can be assumed that differences in signal-to-noise ratio will be reflected in changes in the intelligibility of speech, these data indirectly support the contention of Rutherford (51) and of Krebs (26) that compression preserves the same signal-to-noise ratio as that existing originally. They tend to refute Kretsinger and Young's (27) prediction that a unique masking effect would occur if noise were allowed to enter the compressor along with the speech signal.

The second conclusion drawn from the study is that clipped speech is inferior in intelligibility to both non-limited and compressed speech over a broad range of signal-to-noise ratios when the signal and noise are combined before entering the communication system. This conclusion is supported, once again, by inspection of the data and by the results of statistical analysis. Figure 7 illustrates that the mean speech discrimination scores for clipped speech were poorer than those obtained under the other two experimental conditions at all four signal-to-noise ratios. Statistical analysis showed that performance under the clipped condition was significantly different from either of the other two conditions when the data from all signal-to-noise ratios were pooled.

The mean value obtained for clipped speech in quiet falls well within the range of scores (50 to 90 per cent) reported by those who have investigated the intelligibility of clipped speech. The results across conditions in quiet are in agreement with the findings of Davis, Stevens, and Nichols (15) that compressed speech is more intelligible than clipped speech, but are in conflict with the bulk of earlier findings that clipped speech is more intelligible than non-limited speech in applications involving military communication systems.

The conflict can possibly be resolved by examining the particular conditions under which the latter research was done. It will be re-called that in these experiments radiated power was held constant. Clipping allowed full modulation of the carrier signal by the consonant sounds and, consequently, this full power radiation. This advantage, however, came at the cost of increased distortion. Conventional transmission practices resulted in very little modulation of the consonants and, consequently, very low radiated power for them. It also probably resulted in a substantial amount of masking of the consonants by atmospheric interference.

Within this context, the results of these experiments might better be expressed by the statement that clipped but distorted speech is more intelligible than unclipped speech in which the consonant sounds may be masked and are delivered at a reduced listening level. In short, the circumstances under which these experiments were conducted were sufficiently different from those of the present study as to make the differences in outcome understandable.

The third conclusion drawn from the study is that variation in signal-to-noise ratio appears to affect the intelligibility of clipped speech differently than it does the intelligibility of either non-limited or compressed speech.

This conclusion is supported by inspection of the data and of the outcome of a statistical analysis performed on the data. The analysis showed that the type of amplification x signal-to-noise ratio interaction was significant. Reference to the data graphed in Figure 7 may help to elucidate this outcome.

The graph shows that while the curves representing the performance of the subjects under the non-limited and compressed conditions are very similar, that curve representing their performance under the clipped condition differs substantially in detail. The latter curve is not simply displaced along the abscissa paralleling the performance obtained under the other two conditions at a reduced level of intelligibility. It is, instead, considerably bowed with respect to the other two curves. Specifically, the differences between the curves are greatest under the intermediate signal-to-noise conditions and least under the extreme conditions, particularly at the -5 signal-to-noise ratio, where the difference amounts to only 6.3 per cent.

The deterioration of intelligibility for clipped speech under the +5dB and 0dB signal-to-noise conditions appears to lend support to the report of Kryter, Licklider, and Stevens (28) that when noise is introduced prior to clipping the speech signal appears to "ride" the noise and to be clipped inordinately with a resulting deterioration in intelligibility. This explanation does not appear to hold, however, for a signal-to-noise ratio of -5dB.

Under this unfavorable listening condition the intelligibility of clipped speech was poorer than that of non-limited and compressed speech, just as it had been for the other three signal-to-noise ratios. This tends to refute Krebs (26) contention that when the noise is of greater magnitude than the signal, clipping offers an advantage over compression. Since, however, the difference in intelligibility between clipped speech and the other two conditions was least under this condition and substantially less than the difference observed in quiet, Krebs suggestion cannot be discounted entirely.

The distinct possibility exists that the results obtained at the -5dB signal-to-noise ratio may have been influenced by measurement artifacts, particularly under the clipped condition. The poorest possible score that an individual could achieve on the discrimination task was 0 per cent. Since exactly one-half of the subjects scored 0 per cent under the clipped condition, this could well have artificially limited the range of obtained scores resulting in an inflated value for the mean. A similar, but opposite, effect may have influenced the scores obtained under the quiet condition. Here, subjects could score no better than 100 per cent under the two easier conditions.

In aggregate, the results of the present investigation lead to the conclusion that limiting by compression is the method of choice if compressed speech is equally as intelligible as non-limited speech over a broad range of both favorable and unfavorable listening conditions and under these same conditions clipped speech is consistently inferior.

CHAPTER V

SUMMARY AND SUGGESTIONS FOR FUTURE RESEARCH

Summary

Signal limiting, whether intentionally imposed or not, is an important consideration in every system involved in the transmission, recording or reproduction of sound. In broadcasting, the maximum signal that a station can radiate is restricted by the power handling capacity of the components in the transmitter, as well as by government regulation. In sound-on-film, tape or disk recording, the characteristics of the medium and the saturation level of circuit elements place limits on the magnitude of the signals which can be recorded. In the reproduction of sound, whether in theaters, transportation terminals, or through appliances for the hearing impaired, the maximum permissable output signal level is dictated by the tolerance of the human ear.

There are two commonly recognized methods of limiting. Peak clipping limits by reproducing only that portion of a signal which is below a pre-set level. Any portion of the signal that exceeds this level is not reproduced, but is "clipped." Although clipping provides a simple, effective and inexpensive means of limiting, it also results in the generation of undesirable harmonic distortion because of it's effect on the waveform of the signal.

Compression has replaced peak clipping in most communication systems because it offers many of the advantages of peak clipping without its associated distortion. A compressor operates by reducing its gain instantaneously whenever a signal appearing at its output exceeds a pre-set level. The resulting waveform is merely a reduced replica of the original signal.

Compression limits the absolute magnitude attainable by signals of large amplitude, reduces the dynamic range of a signal, and affords full amplification of small signals below the threshold of compression. These characteristics have been exploited by the broadcasting, recording, motion picture and communications industries. The use of compression also prevents the masking of weaker portions of a signal by boosting them above the level of the background or transmission noise. This results in an improvement in signal-to-noise ratio.

It remains to be seen if the improvement in signal-to-noise ratio made possible in other applications holds true for hearing aid use since the noise problem associated with the use of a hearing aid differs from those problems encountered in other communication systems. In routine hearing aid use noise is present in the listener's environment and enters the amplifying system at the same time as speech.

Although the effect of compression on the intelligibility of speech does not appear to have been investigated systematically under these particular circumstances, several manufacturers of AVC hearing aids have published claims to the effect that the use of these instruments improves hearing in noise. A search of the literature yields only speculation with regard to the effects of limiting on signal-to-noise ratio, and few of the opinions are in agreement.

In view of this conflict of opinion, the lack of experimental evidence, and the potential importance of the matter to the users of hearing aids, the present investigation was undertaken. It was designed to allow comparisons of the relative intelligibility of non-limited, compressed, and clipped speech in quiet, and when noise was allowed to enter the system along with the speech signal.

To achieve this purpose an apparatus was constructed which permitted the combining of speech and noise in any desired ratio. The combined signal and noise were then processed through three separate channels. One of these channels provided non-limited reproduction of speech materials, one provided compressed reproduction, and one provided clipped reproduction. Each channel was terminated in an attenuator used to adjust the level of the signal reaching a subject's earphone.

Eighteen normal-hearing adults served as subjects. They were tested three at a time, with each subject listening to a different experimental condition through one of the three channels. Speech reception thresholds for non-limited, compressed, and clipped spondee words were obtained for each subject in quiet. These thresholds served as the reference values for speech discrimination tests presented at a level of 40dB above each subject's speech reception threshold. The principal test materials consisted of taped copies of Sommerville's recordings of Lists II, III, and IV of the N.U. Auditory Test No. 6 word lists. Speech discrimination scores were obtained for non-limited, compressed and clipped speech at signal-to-noise ratios of -5dB, 0dB, and +5dB, and in quiet, in that order. A different word list was used for every condition of amplification during a single experimental run, and, within each list, a different randomization was used when testing at each of the four different noise levels. The order of presentation of both lists and conditions of amplification was counterbalanced. The subjects recorded their responses on specially-prepared score sheets.

The three speech reception thresholds and the twelve speech discrimination scores obtained for each subject during the conduct of the experiment were treated independently in statistical tests which allowed selected comparisons to be made between any or all experimental conditions and among subjects. The results of the investigation can be summarized as follows:

1. The mean speech reception threshold obtained for compressed speech differed significantly from those obtained for non-limited speech and for clipped speech. The absolute differences observed among all three conditions of amplification were small and probably clinically insignificant. There is some reason to suspect that the observed differences may have been influenced by instrumental artifacts.

2. Compressed speech is equally as intelligible as non-limited speech over a broad range of signal-to-noise ratios.

3. Clipped speech is inferior in intelligibility to both non-limited and compressed speech over a broad range of signal-to-noise ratios.

4. Changes in signal-to-noise ratio appear to affect the intelligibility of clipped speech differently than they do the intelligibility of non-limited or compressed speech. A possibility exists, however, that this finding may have been influenced by measurement artifact.

The experiment may be summarized as having demonstrated that compressed speech is equally as intelligible as non-limited speech over a broad range of favorable and unfavorable listening conditions, and that under these same conditions clipped speech is consistently inferior. It seems reasonable to draw the general conclusion that limiting by compression appears to be the preferred method.

Suggestions for Future Research

The effect that a particular method of limiting may have on the intelligibility of speech is of importance to hearing aid users because hearing aids invariably incorporate some means of protecting the listener from sudden, loud sounds. Consequently, it would seem appropriate to conduct an experiment similar to the present one using hard-of-hearing subjects.

Because it is likely that many of these individuals experience serious difficulty in discriminating speech even under ideal listening conditions, it would probably be necessary to make the listening task easier by eliminating the most taxing listening conditions from the experiment, or, possibly, by substituting a less devastating noise for the cafeteria noise used in the present study.

The question of whether or not clipped speech is, indeed, affected differently than non-limited or compressed speech as a function of signal-to-noise ratio remains to be resolved. Any one of several approaches might be adopted to insure that most scores would fall short of the zero or one-hundred per cent boundaries. For example, a lower sensation level might be used in quiet, or a less disruptive form of noise could be used during the appropriate conditions, or, perhaps, slightly different signal-to-noise ratios such as -3dB, 0dB, and +3dB might be selected.

Finally, it might be appropriate to determine if the effect of method of limiting on speech intelligibility is influenced by such other parameters as restriction of the frequency range or the configuration of the frequency response of the transmission system.

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APPENDIX A

Spondee Words

Alphabetical Listing of the W-2 Spondee Words Used in the Present Investigation for the Determination of the Speech Reception Thresholds

1. airplane
2. armchair
3. baseball
4. birthday
5. cowboy
6. daybreak
7. doormat
8. drawbridge
9. duckpond
10. eardrum
11. farewell
12. grandson
13. greyhound
14. hardware
15. headlight
16. horseshoe
17. hotdog
18. hothouse
19. iceberg
20. inkwell
21. mousetrap
22. mushroom
23. northwest
24. oatmeal
25. padlock
26. pancake
27. playground
28. railroad
29. schoolboy
30. sidewalk
31. stairway
32. sunset
33. toothbrush
34. whitewash
35. woodwork
36. workshop

APPENDIX B

CNC Words

Alphabetical Listing of the CNC Words Comprising Lists II, III, and IV of N.U. Auditory Test No. 6

1. back
2. bar
3. base
4. bath
5. bag
6. bite
7. bone
8. book
9. bought
10. cab
11. calm
12. came
13. cause
14. chain
15. chair
16. chat
17. check
18. cheek
19. chief
20. cool
21. dab
22. date
23. dead
24. deep
25. dip
26. ditch
27. dodge
28. dog
29. doll
30. fail
31. far
32. fit
33. five
34. food
35. gas
36. gaze
37. germ
38. get
39. gin
40. goal
41. good
42. gun
43. half
44. hall
45. hate
46. have
47. haze
48. hire
49. hit
50. hole
51. hush
52. join
53. judge
54. juice
55. jug
56. keep
57. keg
58. kick
59. kill
60. late
61. lean
62. learn
63. lease
64. lid
65. life
66. live
67. loaf
68. long
69. lore
70. lose
71. luck
72. make
73. match
74. merge
75. mess
76. mill
77. mob
78. mood
79. mop
80. mouse
81. name
82. near
83. neat
84. nice
85. note
86. numb
87. pad
88. pain
89. pass
90. pearl
91. peg
92. perch
93. phone
94. pick
95. pike
96. pole
97. rain
98. rat
99. read
100. red
101. ring
102. ripe
103. road
104. room
105. rose
106. rot
107. rough
108. rush
109. said
110. sail
111. search
112. seize
113. shack
114. shall
115. shawl
116. sheep
117. shirt
118. should
119. soap
120. soup
121. sour
122. south
123. such
124. talk
125. tape
126. team
127. tell
128. thought
129. thin
130. thumb
131. time
132. tire
133. ton
134. tool
135. turn
136. voice
137. void
138. vote
139. wag
140. walk
141. wash
142. wheat
143. when
144. white
145. wife
146. wire
147. witch
148. yearn
149. youth
150. young

APPENDIX C

Hearing Aids

Summary of the Attack and Release Times and Compression Ratios of Seven Currently Available Hearing Aids Employing Automatic Volume Control Circuitry

AVC HEARING AIDS

Model	Attack Time	Release Time	Compression Ratio
Acousticon Centennial	5 msec	25 msec	Variable
Audiotone Quietron A 112	20 msec	30 msec	Variable
Fidelity 102 AVC	50 msec	150 msec	Variable
Goldentone Model GII	3.5 msec	30 msec	Variable
Siemens Auriculina	111 msec	333 msec	Variable
Siemens Sirefon Variable	100 msec	500 msec	Variable
Zenith Governor	33 msec	100 msec	5:1

APPENDIX D

The Individual Speech Reception Thresholds Expressed in Decibels Re: 0.0002 Microbar Obtained for Each Subject under Conditions of Non-Limited, Compressed, and Clipped Speech

TABLE 9

THE INDIVIDUAL SPEECH RECEPTION THRESHOLDS EXPRESSED IN DECIBELS RE: 0.0002 MICROBAR OBTAINED FOR EACH SUBJECT UNDER CONDITIONS OF NON-LIMITED, COMPRESSED, AND CLIPPED SPEECH

Subject	Non-Limited	Compressed	Clipped
1	21	23	25
2	19	23	19
3	19	23	15
4	15	21	13
5	19	25	17
6	19	23	17
7	19	23	21
8	21	25	27
9	21	25	21
10	19	21	15
11	21	21	27
12	21	25	25
13	17	19	13
14	23	27	27
15	21	25	27
16	15	17	15
17	21	25	29
18	19	19	29

APPENDIX E

The Individual Speech Discrimination Scores Expressed in Per Cent Obtained for Each Subject under Conditions of Non-Limited, Compressed, and Clipped Speech at Signal-to-Noise Ratios of -5dB, 0dB, +5dB, and Quiet

TABLE 10

THE INDIVIDUAL SPEECH DISCRIMINATION SCORES EXPRESSED IN PER CENT OBTAINED FOR EACH SUBJECT UNDER CONDITIONS OF NON-LIMITED, COMPRESSED, AND CLIPPED SPEECH AT SIGNAL-TO-NOISE RATIOS OF -5dB, 0dB, +5dB, AND QUIET

	-5dB			0dB			+5dB			Quiet		
Subjects	NL	CP	CL	NL	CP	CL	NL	CP	CL	NL	CP	CL
1	6	8	2	50	46	12	80	76	32	98	98	68
2	0	6	6	44	42	16	82	72	28	98	100	80
3	2	0	0	42	16	4	82	68	22	98	92	58
4	2	6	2	48	46	12	88	76	30	98	100	72
5	0	2	0	40	40	12	76	72	12	96	96	52
6	6	0	0	62	32	14	60	76	24	100	100	68
7	12	8	6	58	60	16	86	78	56	100	96	70
8	16	20	0	52	50	6	88	78	30	98	96	68
9	10	4	6	42	50	20	82	86	36	98	98	56
10	16	14	8	52	48	26	86	80	40	100	100	78
11	4	6	0	42	46	14	78	86	30	100	100	74
12	8	2	0	52	44	22	82	72	40	98	100	78
13	10	10	2	64	48	12	92	80	28	100	100	76
14	2	8	0	34	44	10	80	88	16	98	100	68
15	10	4	0	48	48	12	74	70	26	100	100	68
16	12	8	0	60	54	16	78	88	36	98	100	80
17	22	12	4	54	52	22	86	74	32	100	96	78
18	14	18	8	52	52	22	88	74	30	100	98	76

NL = Non-Limited CP = Compressed CL = Clipped