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Intelligibility of Speech Compared Through Two Limiter Compression Circuits

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AN ABSTRACT OF THE THESIS OF Lee M. Odell for the Master of Science in Speech, with emphasis in Audiology/Speech Pathology, presented. February 13, 1974.

Title: Intelligibility of Speech Compared Through Two Limiter Compression Circuits.

APPROVED BY MEMBERS OF THE THESIS COMMITTEE:

Theodore G. Grove

Ronald E. Smith

Hearing aid manufacturers commonly engineer automatic gain control (AGe) circuits which are aimed at reducing'sound tolerance problems and improving speech intelligibility among wearers. The most common type of AGe engineered is one utilizing a fast attack time. The present study was designed to evaluate the effects of both fast and slow attack times on the intelligibility of speech. Twenty-four

normal hearing subjects listened to sixty pre-recorded sentences through two types of hearing aid circuits. Thirty sentences were modified by a fast attack AGe circuit, and thirty sentences were modified by a slow attack AGe. The subjects marked one of four multiple~choice answers for each sentence.

The mean number of sentences answered incorrectly when heard through fast attack AGC was 8.25 . When heard through slow attack AGC, the mean was 6.67. The performance differences which exist between these two modes of signal modification suggest that the fast attack does not improve intelligibility as significantly as slow attack time among normal listeners. Further investigation into the effects of slow attack AGC circuits on the user's ability to understand speech are recommended.

INTELLIGIBILITY OF SPEECH COMPARED THROUGH TWO

LIMITER COMPRESSION CIRCUITS

by

LEE M. ODELL

A thesis submitted in partial fulfillment of the requirements for the degree of

'MASTER OF SCIENCE IN SPEECH

with emphasis in Speech Pathology and Audiology

Portland State University 1974

TO THE OFFICE OF GRADUATE STUDIES AND RESEARCH:

The members of the Committee approve the thesis of Lee M. Odell presented February 13, 1974.

Mary E. Gordon

February 21, 1974

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And to my most loving wife, Peggy, and to my parents, I cannot begin to express my gratitude for their unending patience and encouragements.

PRAISE THE LORD! "And we know that all things work together for good to them that love God, to them who are the called according to His purpose." (Romans 8:28)

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CHAPTER I

INTRODUCTION

Speech as it is encoded by the human ear is a learned perception of variations in two basic physical acoustic phenomena: 1) the rate at which sound waves repeat themselves (i.e., frequency, or Hertz) and 2) sound pressure level (which is related to the intensity or power of the sound waves). An example of a pure tone is graphically represented by the oscilloscope trace shown in Figure 1A. For illustrative purposes this may be contrasted with a typical vowel vocalization in Figure lB. Speech represents a composite of pure tone energies of varying frequency and intensity which results in a complex sound, combining a variety of pure 'tone harmonics. Complex harmonic sounds are generated as air passes through the vocal folds in the larynx. Controlling the size, shape, and use of pharyngeal, oral, and nasal cavities causes sound produced by the larynx to have major resonances at two or three frequencies unique to each speech sound. At these points of resonance, peaks of energy or intensity are created which are referred to as formants (Fletcher, 1953).

Sanders (1971) pointed out that it is essential for the first two formant frequencies to be perceived for proper identification of vowel sounds. Consonant sounds are usually more dependent upon their high frequency components for proper identification, lacking well defined formant regions. Table I indicates the relative phonetic power of

speech sounds as produced by an average speaker. Figure 2 graphically plots the intensities and formants associated with these sounds. It is apparent that much of the acoustic information for consonants lies in the higher frequency region, offering relatively low acoustic intensities when compared to the generally low frequency, higher intensity of the vowel sounds.

The consequences of such frequency-intensity relationships become apparent when one realizes most sensori-neural bearing losses begin in the' high frequency region where the voiceless consonant sounds such as $/f/$, $/p/$, $/s/$, etc. (see Figure 2), are located. The result is a loss in the intelligibility of consonant sounds which may easily become masked by environmental noise such as one may encounter on the street, at a party, or in an auditorium (Denes, 1969; Fletcher, 1953; Sanders, 1971).

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TABLE I

RELATIVE PHONETIC POWER OF ENGLISH SPEECH SOUNDS AS PRODUCED BY AN AVERAGE SPEAKER

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The loss of speech sounds due to an organic hearing deficiency is often overcome through the use of an electro-acoustic amplifying system $(i.e., a hearing aid), which is capable of reproducing a dynamic range$ of sound to include the weakest through the most intense sound pressure levels generated for speech. A conventional linear gain amplifier is commonly utilized for this task. The linear amplifier has constant gain characteristics throughout its designed operating range. For example, a linear amplifier with 50 dB (decibel) gain will amplify a 20 dB input signal to 70 dB output and a 70 dB input signal to 120 dB output. Should the input increase 10 dB, the output likewise would increase 10 dB. There is a systematic relationship between the input and output; any given decibel change at the input should result in a similar decibel change at the output until the output limits of the amplifier are reached. When this happens, the output no longer increases at a rate directly proportional to the input, and the amplifier is said to be entering a state of overload referred to as the amplifier's maximum peak output (MPO) . In other words, the systematic relationship between input and output is no longer linear. As the amplifier reaches MFO, the peaks of the signal are no longer being reproduced. Rather they are being clipped; hence the term "peak clipping" is used to describe this phenomenon (see Figure 3). Peak clipping is a method frequently used in linear amplifiers to limit the maximum sound pressure level at the amplifier's output.

Whenever an amplifier distorts a sine wave, harmonics are introduced into the output signal (Lurch, 1971). The extreme peak clipping seen on the right side of Figure 3 may help to visualize the presence

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Figure 3. An illustration of peak clipping as the amplitude of a sine wave maximum exceeds amplifier output.

of harmonics. The clipped waveform closely resembles that of a square wave, the composition of which consists of sine waves representing the fundamental frequency and, theoretically, all of. its odd harmonics (at. intensities following a mathematical odd harmonic progression). The addition of harmonics to form a composite square wave may be demonstrated by a method known as the "addition of ordinants" (Hirsh, 1952 ; Lurch, 1971; Peterson, 1958; Protter, 1964; Sears, 1961). Figure 4 is an oscilloscope display of a sine wave before and after peak clipping. The oscilloscope vertical gain was adjusted to maintain like amplitudes for purposes of illustration.

Such distortion of speech communication does not present a very serious problem for the vowel sounds since, as can be noted in Figure 2, they are clustered together at the lower to mid-frequencies with high intensity levels. They will not experience much influence from

Figure 4. An oscilloscope display of a sine wave before and after peak clipping. The upper trace is the undistorted sine wave; the lower trace is the sine wave after peak clipping.

harmonics which are both higher in frequency and significantly lower in intensity level. However, the higher 'frequency harmonics of the vowel sounds can interfere with the intelligibility of the more delicate high frequency consonant sounds since the harmonics generated by the vowel sounds may occur within the same frequency and intensity range as the fundamentals of the voiceless consonant sounds. Consequently, the sounds which are already the most difficult to perceive under ideal listening conditions are either partially or completely masked by harmonic distortion.

When such a linear amplifier is used in a hearing aid several other potential problems arise. A second situation is encountered when the output signal of the amplifier is driven beyond the hearing aid user's uncomfortable loudness level (UCL). This' is a particularly important problem if the listener has a hearing pathology which includes a depressed dynamic listening range and a UCL significantly below that

of the normal listener. The third problem is· the amplification of unwanted sounds, which will, in an environment with a poor signal-tonoise ratio, serve as a masking effect for the speech signal. This is a common complaint of persons experiencing sensori-neural hearing pathologies.

In order for an amplifier to reproduce all the phonemes necessary to understand speech in a variety of environments, the amplifier should be capable of amplifying both sounds of very low and relatively high intensities. In other words, the amplifier system should accommodate a wide dynamic range. If the amplifier is to be an effective prosthetic hearing device, it must be capable of amplifying this dynamic range without introducing excessive distortion in the amplified signal. To accomplish this and avoid some of the previously mentioned problems, the linear amplifier has been modified with circuitry often referred to as automatic-volume-control (AVC) or automatic-gain-control (AGC). Although these terms are often used interchangeably, the field of electronic engineering usually makes a distinction according to the design of the amplifier in which the automatic circuitry is incorporated (Lurch, 1971). The term AGC is reserved for the category of amplifier usage which is most likely to include hearing aids.

AGC accomplishes, electro-acoustically, much the same effect as manually rotating the volume control to limit the output of the hearing aid amplifier. The exception to this analogy is that an electronic AGC circuit is much more efficient, faster, and accurate. AGC compression of the output dynamic range helps prevent the amplifier from exceeding

its MFO while at the same time increasing the input dynamic range of the amplifier.

Figure 5 is a block diagram of a representative hearing aid amplifier with AGC circuitry. An input signal (S_i) is fed to the input amplifier, it is amplified, and the amplified signal from the output amplifier $(S_{_{\mathbf{0}}})$ is measured by the detector (AGC circuit) and rectified to a direct current (DC) level which is proportional the output, S_{0} . This DC level (S_f) is fed back to the input amplifier to provide a negative bias, which in turn reduces the gain of the amplifier. In other words, when the output signal becomes sufficiently intense, the AGC circuit introduces a negative feedback, S_f , which reduces the output. Likewise, the detector will also sense a significant reduction in the output signal and within a specified time, as determined by circuit design, the S_f level is turned off, returning the amplifier to a condition of linear gain (Burger, 1970; Carver, 1972; Lurch, 1971; Stuart, 1940).

Figure 5. A block diagram of an amplifier with AGC. The input signal (Si) is amplified by the amplifier and the output signal (S_9) powers the tranducer for the ear. The feedback signal (S_f) reduces the amplifier gain.

Hearing aid amplifiers which utilize AGe circuitry are said to be compression amplifiers. There are three types of compression amplifiers utilized in currently produced hearing aids: A) linear compressors; B) non-linear compressors; C) limiter compressors. The AGC circuit operation described previously is a limiter compressor (Berger, 1970). Figure 6 is a graphical comparison of the output characteristics of the three types of AGe circuitry. Table II is a tabular comparison of the same AGC circuits. Briefly, these circuits differ in their operation

Figure 6. A graphical representation of ideal gain characteristics of the types of AGe circuitry discussed. A) linear compressor; B) non-linear compressor; C) limiter compressor; D) linear amplifier without compression circuitry.

as follows: Curve A) The linear compressor begins its compression action at its very lowest input levels and amplifies at a continuous ratio throughout its operating range. If, for example, it is designed for a 2:1 ratio, an input signal increase of 20 dB will increase the output 10 dB. When the AGC circuitry can no longer reduce the amplifier gain, further increases of the input signal will put the amplifier in a state

TABLE II

TYPICAL OUTPUT CHARACTERISTICS OF THREE TYPES AGC CIRCUITRY INCORPORATED IN A FORTY dB GAIN AMPLIFIER

of MPO. Curve B) The non-linear compressor may have very little compressor action at very low input levels, but as the input signal becomes more intense a continuously greater per cent of the output is compressed until the limits of compression are reached. The amplifier will then proceed to a condition of MPO, but at a higher input than an amplifier without compression or a linear compressor. Curve C) The limiter compressor operates as a linear amplifier without compression circuitry, i.e., at a 1:1 ratio until the AGC circuitry begins reducing the gain. The point at which this occurs is called the threshold of compression (TC) and is commonly designed at a level between 115 and 120 dB (Berger, 1970). As with the previously mentioned types of AGe circuits, the amplifier output using a limiter compressor (Curve C) will also enter a state of MPO as additional input is added after it has reached its limits of compression.

There are two major benefits associated with the limiter compressor as opposed to the other forms of compression mentioned. One involves the ability to provide maximum amplification of the very low intensity sounds and yet limit amplification for high intensity sounds. The other advantage is a greater dynamic range of operation before the limits of compression are exceeded (see Figure 5).

A disadvantage to limiter compression (Curve C) is its inability to react instantaneously. There is typically a 4 millisecond (ms) to 50 ms time delay in attenuation after the sound has attained the threshold of compression. This onset delay is called the "attack" time. . Likewise, there is a time delay from the point at which a compressed signal is reduced below the TC until the AGe releases the control of the gain. This phenomenon is referred to as the "release" time.

Figure 7 is an oscilloscope envelope display of a typical audio signal which has been subjected to limiter compressor action. Beginning at time zero is a normal uncompressed steady-state signal. The time interim between 1 and 2 is the "attack" time. The distance the signal extends above the steady-state AGe level is the "overshoot." The interim between 2 and 3 represents the time necessary for the AGC to reach its steady-state level after the "attack" and is called the recovery time. Point 3 represents the beginning of steady-state AGe operation. At point 4 the signal is reduced to below the TC. It may be noted at this point in Figure 7 that the AGC has not yet released its control of the signal, so that the smaller signal is also compressed for a small period of time. The release time is represented by the time interim between points 4 and 5.

Figure 7. An oscilloscope envelope display of a typical audio signal which has been subjected to limiter compressor action. 1 to 2 is the attack time, 2 to 3 is the recovery time, and 4 to 5 is the release time.

Those phases represented in this graphical illustration which are considered to be of most importance to hearing aid designers are the attack and release times. The attack time of the AGC unit must be longer than the time necessary to complete one full cycle of the lowest frequency that the amplifier is designed to pass. Most commercial hearing aids available today have attack times of 50-ms or less. If the attack time is not long enough, the AGC circuit would interpret the rise time of each low frequency sine wave as the onset of separate signals rather than measuring the content of the signal as a whole. This can result in a distorted signal, which would sound like a flutter if the recovery time were also relatively short. Because of the relatively slow rate at which syllables are uttered (100 to 150 ms average), release times commonly range from 50-150 ms in hearing aid circuits (Berger, 1970; Carver, 1972; Rintelman, 1972).

CHAPTER II

HISTORY

In the early history of radio broadcasting, volume control for the transmitted signal was accomplished manually. When the program material became too loud, a technician had to lower the volume and as the sound level returned to normal the volume had to be increased again. Particularly disturbing were sudden loud noises. The technician operating the volume control had extreme difficulty acting rapidly enough to reduce the sound level before the members of the listening audience were elevated from their seats. What was needed was a robot with inhumanly fast reaction time, continuously alert to the changing program material, and never tiring of his job.

By the late tbirties and early forties a method of negative feedback was implemented to limit the output of such amplifying circuits (Black, 1941; Cook, 1939; Stuart, 1940). The first electro-acoustic hearing aids were produced around the turn of the century, and by the time methods of automatically limiting gain were developed, it seemed apparent this type of circuitry might have some utility for hearing aid users also (Davis, 1947 ; Hudgins, 1948). The difficulty was that the state-of-the-art in electronics was insufficiently developed to produce a limiter circuit small enough to fit into a wearable hearing aid.

A study reported in 1948, by Hudgins, et al., comparing a master hearing aid and an experimental hearing aid, both equippped with AGC

circuitry, to two commercially available hearing aids without compression circuitry, indicated the hearing impaired person would derive benefit from utilizing AGC. One of the first reported uses of automatic gain control in a commercially available hearing aid was produced by a European manufacturer. This was designed to be used as a nonportable desk type amplifier and was larger than many of today's dictating recorders (Berger, 1970; Poliakiff, 1950; Caraway, 1966).

Commercially produced, wearable hearing aids with automatic gain control began to appear on the market in the United States in $1949.$ It was not long before more critical testing and experimentation began, in order to evaluate how_more effectively compression circuitry improved speech intelligibility than-conventional linear amplifiers using peak clipping to limit maximum output (Edgardh, 1952; Parker, 1953). The results were impressive and supportive in favor of the compression type circuitry.

Later, in 1960, Kretsinger and Young reported a study comparing two degrees of fast limiting compression (10 dB and 20 dB) to peak clipping of the same degrees to evaluate their relative effect on intelligibility of speech. As in the earlier studies, it was apparent that using peak clipping to restrict the maximum output would limit the listener's ability to achieve good intelligibility scores more than either of the compression limiters.

In 1963, Lynn and Carhart reported on a study in which a variety of attack and release times were compared. Due to the large number of compression limiters being built and used in hearing aids, such study was not unreasonable. The study utilized nine circuits which compared

attack times ranging from 5 ms to 85 ms and release times from 30 ms to 1200 ms. A conventional fixed gain amplifier was also utilized as a basis for comparison. Two measures of effectiveness were used. The first incorporated speech reception threshold measurements as recorded by hearing threshold dial readings, and the second measured intelligibility using Phonetically Balanced (PB) words. The subjects were divided into three groups: those with pathologies of otosclerosis, labyrinthine hydrops, and presbycusis. It was concluded the otosclerotics received minimal benefit from compression regardless of the time constants. The remaining subjects appeared to derive significant benefit from the use of the limiter compression amplification with maximum results occurring when attack and release time constants were about 5 ms and 150 ms, respectively. The authors of the study emphasized that primary consideration must be given to the user's needs, the type of compression system used, and the levels and varieties of sounds he might encounter in his environment.

Statement of the Problem

studies to date have been concentrated in the area of evaluating attack times for fast limiters. There is no available information indicating how speech intelligibility, when assessed with a sentence discrimination task, is affected by a limiter compressor which utilizes long attack and release times. There is the possibility of improved performance through the utilization of attack times which are more than 500 ms. The purpose of this study is to compare the intelligibility of speech through an amplifier with a fast attack time and a slow attack , time utilizing a sentence discrimination task.

CHAPTER III

METHODS AND PROCEDURES

It is the intent of this study to compare two rates of limiter compression and study (analyze, etc.) their relative effects on speech intelligibility using a sentence discrimination task. The hypothesis may be stated as follows: There is a difference in intelligibility of speech when compared through fast and slow attack AGC circuits. The methods and procedures used to conduct this study are outlined below.

I. SUBJECTS

Twenty-four normal hearing young adults were selected to serve as subjects for this experiment. There was no attempt to balance the subjects according to sex. In order to control for possible effects of age or auditory pathologies, subjects were chosen according to the following criteria: 1) normal hearing as determined by pure tone thresholds of 10 dB (ANSI 1969) or better at frequencies of 250, 500, 1000, 2000, 4000, and 8000 Hertz; 2) ages from eighteen to thirty years; 3) no previous history of excessive noise exposure; 4) a negative history of hearing pathologies.

II. TESTING ENVIRONMENT

All testing was conducted in single-walled Industrial Acoustics Company (lAC) sound treated rooms located in the Veterans Administration Hospital Audiology Service, and the Portland State University Speech , and Hearing Sciences Audiological Testing Environment, Portland, Oregon $(models 404 and SP 403, respectively)$.

III. DISCRIMINATION TEST

The Harvard University Psychoacoustic Laboratory (PAL) sentence discrimination test number 8 (PAL-8) was utilized. Two studies (Jerger, 1966_a; Jerger, 1966_b) indicated the PAL-8 test material had superior capabilities for ranking and ordering hearing aids when compared with the use of phonetically balanced (PB) monosyllabic word discrimination tests. Also, since most hearing aid usage involves listening to continuous discourse, it was deemed more relevant to use a sentence discrimination task rather than PB words. Table III shows two examples of typical sentences used for the discrimination task. These sentences have been constructed in such a manner that the listener must hear most or all of the key words before he can derive the appropriate answer.

TABLE III

SAMPLE SENTENCES FROM PAL-8 TEST

1. What insect does honey come from? hive bee cricket treasury

2. Underline the smallest sum of money: dwarf flower 5 cents mouse

The test consisted of sixty short sentences which were in the form of questions, commands, or incomplete statements. Each subject was instructed to underline the most correct multiple choice response following presentation of the sentence. A complete copy of the test material utilized for the experiment is located in Appendix A .

IV. EQUIPMENT

Threshold Measurements

All pure tone threshold measurements were obtained with a Grason and Stadler Model 1701 dual channel automatic audiometer with TDH 49 earphones mounted in MX 41/AR cushions (V.A. Hospital) and a Beltone 15C audiometer with TDH 39 earphones, also mounted in MX 41/AR cushions (PSU). Each audiometer was monitored for correct calibration with a Bruel and Kjaer sound level meter Model 2203 fitted 'with a thirteen octave band acoustic filter and artificial ear with a 2 cc coupler before and after testing.

AGC Limiter

Two experimental automatic gain control circuits were incorporated in a hearing aid amplifier. The amplifier input and output were modified in such a manner as to match. the higher intensity and impedance of the tape recorder output and input, respectively. This engineering represents a necessary alteration in the typical hearing aid amplifier, which has its input and output impedances to match a microphone and receiver, respectively. A schematic wiring diagram of the apparatus is included in Appendix B.

The fast limiter was designed to produce an attack time of 40 ms and the slow limiter an attack time of 600 ms. In order to limit any

release time variables in the study, both limiters were designed to produce the same time constants of 650 ms. A Tektronix 5310 dual beam oscilloscope was utilized to measure the actual time constants of the AGC amplifier. The activating signal was a 1000 Hertz sine wave.

Stimulus Tape Recording

The master recording of the PAL-8 test stimuli was made in the \ LAC Model 404 audiometric test booth described in the testing environ ^Iment section. Sixty PAL-8 sentences were read into an AKG Model D200E dynamic studio recording microphone connected to a Teac 70308L tape recorder. Maxell TID 50 extended range high fidelity recording tape was the medium for recording. The sentences were read at ten-second inter vals. This procedure resulted in an unmodified recording of the sentences (see Figure 8). The recorded list of sixty sentences was then: 1) routed through the hearing aid circuit modified by the slow attack AGC and dubbed onto track 1 of another tape and 2) routed through the hearing aid circuit modified by the fast attack AGC and recorded on track 2 of the other tape. The dubbing tape recorder which recorded the AGe modified signals was a Tandberg 4000X tape deck.

During the dubbing process the AGC amplifier gain was adjusted to provide the amplified running discourse intensity peaking at approximately 8 dB above the threshold of compression. The AGC amplifier input and output were continuously monitored by a Tektronix Model 5310 dual beam oscilloscope to ascertain that proper gain and compression relationships were maintained.

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Figure 8. Block diagram showing the procedure utilized to develop composite test presentation tape.

The final recording used for the stimulus test tape was dubbed from the above described tape and divided into four forms: A, B, C, and D. Forms A and B had sentences 1 through 30 modified by slow attack and sentences 51 through 60 modified by fast attack. Form A presented the slow attack sentences (1-30) first and Form B presented the fast attack sentences (31-60) first. Forms C and D reversed the order of modification. Sentences 1 through 30 were modified by the fast attack AGC. Forms C and D also were arranged so the fast and slow attack sentences were each presented first, as they were for Forms A and B.

During the production of the final recording just described above, a signal mixer built into the dubbing recorder (Tandberg 4000X) was utilized to introduce a masking noise which would provide a difficult listening task for normal hearing persons. "Speech Noise," which is a random noise generated and filtered by the Grason Stadler 1701 audiometer, was the masking utilized for this study. The acoustic characteristics of the speech noise are graphically displayed by Figure 9 (Grason stadler 1701 operating manual).

A pilot study involving ten normal hearing persons suggested that a -4 dB signal-to-noise ratio (the level of the signal was 4 dB below the intensity of the. noise) would result in scores of approximately 65% when mixed with a signal not modified by AGC circuitry. The -4 dB signal-to-noise ratio hence was adopted for the study being conducted.

The net result of producing the test stimulus recording was four forms of stimulus presentation $(A, B, C, and D)$, all originating from the same master recording. Irregularities which may have existed in the master recording were represented equally in each mode of signal modification thus balancing effects across treatments. Likewise any order effects, which may have developed during stimulus presentation, and which could have enabled a subject to answer the second thirty sentences with greater accuracy than the first thirty sentences, also were balanced across treatments. Table IV displays the stimulus presentation order of Forms A , B , C , and D . Six subjects were assigned to each of the forms $(A-D)$. Form A was presented to subject 1 (S_1) , Form B to S_2 , \ldots , and Form D to S_{24} .

TABLE IV

STIMULUS PRESENTATION ORDER OF FORMS A, B, C, AND D

CHAPTER IV

I. RESULTS

The hypothesis that there is a difference in intelligibility of speech when compared through fast and slow attack automatic gain control (AGC) hearing aid circuits was supported by the data. Figure 10 represents a composite of histograms which display various aspects of these data. Histogram A expresses the mean number of group errors and standard deviations for the fast and slow attack modes. The mean intelligibility error score for the slow AGC group was 6.67 out of a possible total of 30 correct responses, and for the fast AGC group the calculated mean was 8.25. The differences between mean intelligibility error scores for fast and slow attack AGC circuits were statistically significant at the .05 level of confidence. Histogram B represents an analysis of error scores of male and female subjects within the experimental groups. Females scored lower on the PAL-8 intelligibility test, irrespective of AGC mode, than males, although the differences,between means, 7.93 errors for females and 6.98 for males, were not statistically significant. Histogram C reflects the means and standard deviations of the reversal design in order of sentence group $(1-30)$ and 31-60) presentation. The mean number of intelligibility errors between the sentence group presented first, 7.67, and the sentence group presented second, 7.25, did not reveal any differences that could contribute toward an order effect.

Figure 10. Histograms expressing mean error score on the PAL-8 intelligibility test, according to A) AGC mode, B) sex of subjects, C) presentation order of sentence groups (1-30 and 31-60).

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Since the study necessarily was conducted in two separate testing environments, Portland state University and Veterans Administration ! Haspital, it was of interest to examine the error scores of the two' samples to determine whether the experimental settings were autonomous. The means for the university and V.A. Hospital samples were quite similar, 7.39 and 7.55, respectively, with standard deviations of 2.90 and 3.0, indicating that separate test environments did not contribute significantly to the outcome 'of the study.

Finally, the age range of the total experimental sample assigned to the various conditions was 19 to 30 years. No attempt was made to match experimental subgroups on the basis of age; consequently, it was of additional interest to determine whether age was a determining factor in the intelligibility error score on the PAL-8 examination. A Pearson Product Moment correlation (Bruning, 1968) was computed between age in years and number of errors on the test. An extremely low correlation between these two variables $(r = .11)$ suggests that age and error score were not related within this restricted sample.

II. DISCUSSION

1) An evaluation of the data generated by this study indicates the emergence of a concept which may be the converse of that practiced by the hearing aid industry as a whole.

2) The general trend in AGC circuit design is in the direction of reducing the attack times (Berger, 1970; Carver, 1972; Rintelman, 1972). 3) Apparently, some companies are attempting to develop AGC circuits for hearing aids with attack times less than 5 ms (Hewitt, 1972). This trend toward faster attack times is based on the supposition that the sooner the AGC can become effective after the onset of an excessively intense speech signal, the sooner the peak clipping and harmonic distortion will be eliminated. Accordingly, one would anticipate a resulting speech signal which could be-more easily understood. The results of this study indicate the inverse may be true, at least within the limits of the amplifier defined in Appendices B and C. This disparity may be better understood through examination of the experimental AGC amplifier operation.

A study of the acoustic dynamics of the experimental AGe ampli-. fier in operation, revealed the attack circuits reduced the gain much slower for speech signals than for a reference pure tone signal of 1000 Hz. The actual lapsed time the AGC circuits required to achieve full AGC control when subject to experimental test stimuli was measured. The fast circuit had an average lapsed time, from signal onset to full AGe control, of 110 ms. For slow attack the mean lapsed time was 1800 ms. Comparing these figures with those obtained using the reference pure tone signal, the fast AGe circuit was rated at 40 ms and the slow

circuit at 600 ms. The significance of this greater lapsed time is that slow AGC may provide better amplified listening conditions than fast AGC for the signal levels utilized in this study. This can be visualized by examining the effect of these modes on signal intensity.

The difference in signal amplitude $(3$ to 5 dB) between a condition of fast AGC and slow attack prior to AGC control may be sufficient to produce a significant difference in speech discrimination scores. In marginal listening situations, which were defined in this study by a high level of masking noise introduced after AGC, the non-peak areas of the &ignal may be more clearly heard through the slow attack cireuit than through the fast attack circuit for the period of time the difference exists. This difference between the fast and slow attack outputs may be more clearly visualized if the signal characteristics are analyzed more closely.

Figure 11 shows a direct comparison of an idealized pure tone signal (11A) and a typical speech signal (11B). Figure 12 shows the pure-tone signal before and after peak clipping (12A and 12B respectively). The portion *of* the sine wave which has undergone peak clipping represents a large proportion of the horizontal width (time) of the signal. Figure 13 provides an example of the typical speech signal before and after peak clipping, but prior to AGe control of the signal. The center oscilloscope trace is the undistorted signal without peak clipping; the upper and lower traces represent the signal undergoing peak clipping. It may be observed that only the very largest peaks are being clipped. These are very short in time duration, i.e., their horizontal width is very narrow when compared to the width of one cycle of

! .

Figure 12. A) Pure tone sine wave undistorted. B) Pure tone sine wave with peak clipping.

Figure 13. An example of a typical speech signal being peak clipped. The center trace is the undistorted signal. The upper and lower traces are the same signal after peak clipping.

the fundamental frequency.

The above information indicates that unless the amplifier is overdriven so that the signal is more severely peak clipped than it was for this study, the harmonic distortion introduced through the clipping of the narrow peaks may not significantly deteriorate the overall signal quality. Inspection of Figure 13 seems to support this. It can be noted that there has been very little harmonic distortion added to the original signal. This is not to imply the particular condition presently being discussed may be generalized to all other listening environments. The listening conditions presented for this study, for experimental purposes, include only a three-second speech stimulus without any non-related acoustic interaction prior to AGC modification of the signal.

In order to eliminate independent variables from having an unknown effect on the test results, this investigation was conducted in an acoustically sterile environment, quite unlike a'hearing aid user might encounter in everyday life. Most speech signals, when compared to the one used in this study, have much variability in rate and intensity. The speech stimulus utilized for the experiment was presented at a steady rate and peaked at a consistent level (zero dB on a $V.U.$ meter). Sterility of the acoustic environment was further assured by insuring a low background noise level during the recording. A compression type amplifier in a hearing aid is most useful and effective to the user when in an environment with a moderate ambient noise level. A more pragmatic test situation would include introduction of the masking noise with the speech stimulus prior to the AGC circuit.

Finally, the difference in intelligibility scores for the two modes of AGC also may be attributed to the comparatively abrupt change of signal modification of the fast AGC. The transients of the fast AGe might be sufficiently detectable to the subject to impede his maximal ability to discriminate speech sounds. Figures 14 and 15 are actual oscilloscope envelope displays for the AGC circuits utilized for this study. The circuits are being subjected to a sudden 20 dB sine wave signal which is greater than the TC . Figure 14 displays the abruptness of the signal as it responds to fast attack AGC control. Figure'15 reveals the more gradual envelope of slow AGC response. A speech signal subjected to this form of signal modification is less likely to cause discernible distortion due to AGe transients.

Figure 14. An oscilloscope,envelope display of the fast attack AGC operating characteristics when activated with a 20 dB sine wave pulse.

Figure **15.** An oscilloscope envelope display of the slow attack AGe operating characteristics when activated with a 20 dB sine wave pulse.

CHAPrER V

CONCLUSIONS AND IMPLICATIONS

I. SUMMARY

The focus of this investigation was directed toward a better understanding of the effects of hearing aid automatic gain control (AGC) on the intelligibility of speech. Specifically, the study compared a fast attack AGC circuit with a slow attack AGC, whereas all other variable parameters associated with AGC amplifiers were held constant.

A hearing aid AGC amplifier was modified for this study to provide a fast attack time of 40 milliseconds (ms) and a slow attack time of 600 ms. The release time was 650 ms for both attack circuits. Twenty-four normal hearing subjects (18-30 years Old) listened to sixty pre-recorded PAL-8 sentences. Thirty sentences were modified by fast attack AGC and thirty by slow attack AGC. Each subject had a printed form which included four multiple-choice answers for each sentence. The most correct answer was to be underlined. In order to provide a sufficiently difficult task for normal hearing subjects a masking noise was dubbed onto the stimulus tape recording after the hearing aid AGC output was recorded at a -4 dB signal-to-noise ratio.

The mean number of sentences answered incorrectly when heard through fast attack AGC was 8.25 , and when heard through slow attack AGC the mean was 6.67 errors.

II. CONCLUSIONS

The results of this investigation justify the following conclusions:

- 1. Differences of intelligibility between fast and slow attack AGC were statistically significant at the 0.05 level of confidence when conducted under the conditions of this study.
- 2. The slow attack time mode resulted in greater intelligibility of speech than the fast attack time mode.
- ¹ 3. The AGC attack times should be assessed with a typical speech signal as well as a pure tone signal. The possibility exists that the test stimulus duration may be inadequate to derive full benefit of the AGC. A speech discrimination task utilizing single word presentations is possibly too short in time duration to test AGC capabilities.
- 4. The PAL-8 sentence discrimination test is a satisfactory tool for hearing aid evaluation. The test should be a particularly well-suited tool if the hearing aid utilized AGC. Due to the vintage of the material (1944) and the population for which it was intended (military) some sentences may have to be edited because of a lack of knowledge of the subject.

Other considerations may be implied for future study as a result of the present study:

- 1. Conduct a study in which masking noise is introduced prior to AGC. This would provide information about AGC operation in circumstances more consistent with \ those found in everyday hearing aid usage.
- 2. The use of subjects with hearing pathologies would be important since, typically, hearing aid users do not have normal auditory acuity.
- 3. A variety of AGC attack times, other than 40 ms and 600 ms, should be investigated.
- 4. The effect which a varying speech stimulus intensity has on speech intelligibility should be investigated. This condition would be more comparable to conversational speech than the closely regulated intensity used for this study.

In addition to the implications suggested above, the hearing aid evaluation (HAE) is an aspect of clinical practice in which knowledge derived from this study may be applied. Evaluation of hearing aids on' patients should consider the aids' AGC characteristics and the duration of the speech discrimination stimulus utilized. Further, the clinician should consider the use of background noise as an integral portion of the RAE procedures. An understanding of AGC characteristics when subjected to speech signals should add further impetus to the consideration of background noise utilization for BAE's.

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APPENDIX A

MASTER TAPE RECORDING SCRIPT AND SUBJECT TEST FORM

This is a test to see how well you can hear sentences spoken in noise. Some of the sentences are questions or commands. Other sentences are incomplete statements. After every sentence, you are to circle or underline the word or number which answers the question or command, or which completes the sentence. ALWAYS respond to each sentence. If you are not sure, GUESS. DO NOT LEAVE ANY SENTENCES UNANSWERED.

Here are some practice sentences. Number one has already been correctly marked. ARE YOU BEADY?

- 1. Great Lakes 3. 12 Atlantic 1 Pacific 9
Mediterranean 7 Mediterranean 7
- 2. April 4. Man $01d$ Mountain Mo Mountain Idea Ant Seal Mouse

That is how the test will go. ALWAYS make a mark, even if you have to guess.

TURN THE PAGE OVER & WAIT FOR TEST TO BEGIN.

1. Litter Ladder Letter Latter

2. 12 4 8 3

3. Alaska Ice Eskimo Mines

- 4. \mathbf{I}_1 2 8 1
- $5. 2$ 35 7 12
- 6. Ocean Rotten Food Broken-down
- 7. Furnaces Winter Tickling Spring
- 8. Leg Body Head Foot
- 9. Kindness Slow Runners Food

10. 10 A.M. Lunch Supper 11 P.M. 11. 7 4 Car Umbrella 12. Hot Summer

- Thermometer stove 13. Soldier
- Bully Cold Boxer

 14.6 Yes No 11

- 15. Captain Major Minor Corporal
- $16.$ August .october November Autumn
- 17. Sunday England Washington Rome
- 18. Silk Fish Caterpillar Worm-hole
- 19. 30 38 25 72 $20.$ $10\frac{1}{2}$

5 20 10

Ice Mines 22. Pipes Fire Windows Leaves

21. North-Pole Winter

23. Donuts Soup Alcohol Wasp

24. Den Sheep Dog Bark

25. Lifting Burning Traveling Flying

26. No White Yes Winter

27. Music Tennis Shooting Loud Noise

28. Keyhole Mattress Mat Floorwax

29. Hive 'Bee Cricket

30. Blind Glasses Dark

Treasury

turn the page

Invisible

-
- 2. Round 12. Yes
Steel Mt. Glass
-
- 4. Diamonds 14. Gum
Blood Gun Blood Gun
Green Bun Green Bun
Sky Bum Sky Bum
- 5. Ground 15. Skating
Stoves Reeping
- 6. Gray 16. Meals
- 7. 13 17. Meals
17 Break:
- 8. Dwarf 18. 4 Flower 75. 5 cents 2 Mouse 50
- First
Hurdler

Press West Paper Red

- 1. Round 11. Swindle
Red Mist Red Mist
Sweet Tadp Sweet Tadpole
Cider Puppy Puppy
	- Steel Mt. Everest Drinking-water
No
- 3. Forks 13. Compass

Lassoes Printing Lassoes Printing-press

Hooks Author Author Food Feet
	-
	- Keeping money Cellars Fur coats
North-Pole Getting m Getting married
	- Sunny Fire-engines Blue Mailman Picnic Envelopes
	- **Breakfast** Baby Supper 2 ·Weeks Lunch
		-
- 9. Referee 19. Diamonds
First Lead Expensive Last Elephant
- 10. News 20. Evening Reporter Chair
- 21. Eating Garden Caterpillars Fishing
- 22. Kitchen Fishing Beautiful Library
- 23. Locomotive Engineer Tracks Coach
- 24. Moving companies Actors Film Canvas
- 25. Admiral King Popeye General MacArthur Greta Garbo
- 26. Grapes Fruit Sherry Alcohol
- 27. Night Beds Rest Meals
- 28. Giant Mouse Man Dwarf
- 29. Texas Platter Lake Superior Rhode Island
- 30. Yes 5 No 10

turn the page

RECORDING SCRIPT AND ANSWERS FOR PAL-8 SENTENCE

SPEECH DISCRIMINATION TASK

This is a test to see how well you can hear sentences spoken in noise. Some of the sentences are questions or commands. Other senter Some of the sentences are questions or commands. Other sentences are incomplete statements. After every sentence, you are to circle or underline the word or number which answers the question or command, or which completes the sentence. ALWAYS respond to each sentence. If you are not sure, guess. DO NOT LEAVE ANY SENTENCES UNANSWERED.

Here are some practice sentences. Number one has already been correctly marked. ARE YOU READY?

That is how the test will go. Always make a mark, even if you have to guess.

TURN THE PAGE OVER AND WAIT FOR THE TEST TO BEGIN.

ARE YOU READY? Circle Form A (B, C, D)

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TURN THE PAGE

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James Barnett
James Barnett

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APPENDIX B

AMPLIFIER SCHEMATIC

APPENDIX C

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DATA ON AMPLIFIER CHARACTERISTICS

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APPENDIX D

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RAW DATA

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THIS THESIS WAS TYPED BY

LESLIE GRUEGER.