Comparative study of analog and digital hearing aids

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COMPARATIVE STUDY OF ANALOG AND DIGITAL HEARING AIDS

A Thesis

Submitted to the Graduate Faculty of the
Louisiana State University and
Agricultural and Mechanical College
in partial fulfillment of the
requirements for the degree of
Master of Arts

in

The Department of Communication Sciences and Disorders

by
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B.A., Louisiana State University, 1998
May 2002

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ABSTRACT

The purpose of the present study was to determine if objective and/or subjective differences between analog and digital hearing aids exist when blinding is utilized in the protocol and circuitry is controlled. Ten normal hearing and seven hearing impaired subjects were monaurally fitted with analog and digital hearing aids. Probe microphone measures were obtained at the plane of the tympanic membrane at two output levels (40 dB SPL and 70 dB SPL). Listener performance in quiet was evaluated via word recognition testing, listener performance in noise was evaluated via the Hearing in Noise Test, and listener preference was evaluated via a questionnaire. Results indicated similar performance for all objective and subjective tasks for both hearing aids with the exception of better performance in quiet at the 40 dB SPL presentation level with the analog hearing aid for the hearing impaired group. These results indicate that listeners performed as well or significantly better with the analog hearing aid than with the digital hearing aid. Furthermore, future investigation is recommended to evaluate the effectiveness of some features available on digital hearing aids that are not available on analog hearing aids, such as expansion and noise reduction.

CHAPTER I

INTRODUCTION

Digital hearing instruments sales are increasing dramatically. A dispenser survey (Strom, K.E. 2001) reported digital hearing instrument sales increasing from 6% in 1996 to 23% in 2000 for all dispensing professionals polled. In the same survey, average retail cost to consumers for a single digital hearing instrument was reported as being $2496.00 compared to the $1618.00 charged for an analog programmable hearing instrument. Unfortunately for the consumer, after digital hearing instruments have been on the market for several years, the question is still being asked: "Does digital processing *per se* produce superior produce superior speech perception scores, independent of the diverse speech-processing strategies and architectures that different hearing aids employ?"(Ross, M. 2001)

Previous studies comparing digital and analog hearing aids have demonstrated no definitive objective advantage favoring digital hearing aids and have shown only mixed subjective preference between the two types. However, previous studies reviewed comparing digital to analog hearing aids did not control for circuitry type. Therefore, it is possible that differences between analog and digital performance evident in past studies may have been attributed to circuitry differences between the hearing instruments rather than to the different means of signal processing. In addition, several of the previous studies did not utilize blinding in the design of the study. Therefore, it is possible that the subjective results of these past studies may have been confounded due to the Hawthorne effect (Valente, M., Fabry, D.A., Potts, L.G., Sandlin, R.E. 1998; Bentler, R.A. 2000).

Thus, the purpose of the present study was to evaluate listener performance with analog and digital hearing aids when circuitry was controlled and blinding is utilized in the protocol.

CHAPTER II

REVIEW OF LITERATURE

INSTRUMENTATION

Analog Instruments

Analog hearing instruments consist of a microphone, a pre-amplifier, a means processor, an amplifier and a receiver. During analog processing, the microphone transduces the acoustic input signals into electrical input signals. The pre-amplifier amplifies the electrical input signals and the means processor spectrally shapes the frequency response. After spectral shaping, the amplifier amplifies the electrical signals, which are then transduced by the receiver into acoustic output signals (Holube, I. and Velde, T.M. 2000).

In analog hearing instruments, both the acoustic and electrical signals are continuous in time and in amplitude. Between any two moments in time, there are an infinite number of instants the signal exists and, at any single moment in time, there are an infinite number of possible amplitude values of the signal (Rosen, S. and Howell, P. 1991).

Analog hearing instruments may be digitally programmable, however signal amplification is still accomplished via analog means. Digitally programmable analog hearing instruments allow settings such as frequency response and gain to be manipulated digitally using a computer or hand-held programmer, however, digitally programmable analog hearing instruments do not provide true digital signal processing (Venema, T.H. 1998).

Digital Instruments

Digital hearing instruments consist of a microphone, a pre-amplifier, an analog-to-digital converter, a digital signal processor, a digital-to-analog converter, an amplifier and a receiver. During digital signal processing, the microphone transduces the acoustic input signal into an electrical input signal. The electrical input signals are amplified by the pre-amplifier and are digitized by the analog-to-digital converter. The digital signals are spectrally shaped by the digital signal processor and are converted into analog electrical signals by the analog-to-digital converter. The electrical signals are then amplified by the amplifier and transduced into an acoustic output signal by the receiver (Lybarger, S.F. and Lybarger, E.H. 2000).

In digital hearing instruments, neither the acoustic nor the electrical signals are continuous in time and amplitude. Between any two moments in time, there are a discrete number of instants the signal exists and, at any single moment in time, there is a limited set of possible amplitude values of the signal (Rosen and Howell, 1991). Stated differently, the input signal is sampled at discrete points in time and each sample is truncated or rounded to a specific quantity within a discrete set of values (Holube, I. and Velde, T.M. 2000).

Purported Advantages of Digital Hearing Instruments

Signal shaping in digital systems is performed with numbers under control of algorithms that provide extremely precise signal changing calculations. Analog systems may exhibit differences in performance due to slight variations in the ability of the electrical components to reproduce the signal, which cause discrepancies in the production process. Digital systems exhibit consistent performance in the production

process through precise algorithms, thereby resulting in greater fidelity of the signal (Holube, I. and Velde, T.M. 2000). In addition, filter specifications in analog systems become less accurate as components age. In digital systems, however, the exact numeric representation of the input signal throughout the process of digitization provides consistent performance over time (Lunner, T., Hellgren, J., and Arlinger, S. 1993).

Digital hearing instruments provide additional advantages over analog signal processors in the form of greater reliability over time, the robustness of binary code, decreased circuit noise, and greater fitting flexibility because of the ability to program the digital instrument to fit steep or unusual hearing loss patterns (Schweitzer, C. 1998; Preves, D.A. 1995).

Features of Digital Hearing Instruments

Hearing instruments with true digital signal processing have features that are not available in hearing instruments with analog signal processing. Features that are currently offered in digital instruments that are unavailable in analog instruments are noise reduction, feedback reduction, and expansion.

Noise Reduction

Excessive background noise is a common complaint among hearing aid wearers and proves to be one of the most difficult problems for hearing aid manufacturers and dispensers to address (McFarland, W.H. 2000; Sandlin, R.E. 1995). Noise reduction technology in hearing instruments attempts to improve listener comfort and increase speech intelligibility by reducing the effects of background noise. In current analog instruments, the only ways to decrease the prominence of background noise in the

amplified signal is to either decrease the low frequency amplification (which inevitably reduces the amount of amplified information the end user is receiving), or to employ a directional microphone (Kennedy, E. 1997). The use of a directional microphone has been shown to improve intelligibility of speech in noise without decreasing the amplification of low frequency information (Killion, M.C. 1997). In digital instruments, the same strategies can be employed as in the analog instruments, but these instruments can also make use of a noise reduction algorithm. A noise reduction algorithm is a computer program within the hearing aid that reduces amplification in the frequency region where steady-state acoustic modulations (such as the constant babble of speech noise) are present in the processed signal. One drawback to this type of noise reduction is that desired signals with frequency modulations similar to noise, such as music, result in undesired gain reductions (Venema, T.H. 1998). In fact, distortions in the speech signal introduced by noise reduction processing may be more deleterious to intelligibility than background noise (Levitt, H. 1997).

Feedback Reduction

An additional feature available in digital hearing aids is feedback reduction. Acoustic feedback is a squealing sound that can be bothersome to the hearing instrument user and/or the conversation partners. Acoustic feedback is caused by the re-amplification of output signals from the receiver (Preves, D.A. 1995). In analog hearing instruments acoustic feedback is controlled by decreasing the gain of certain frequency regions. This decrease in gain can degrade the output signal and reduce intelligibility. Another technique is to employ a narrow-band filter at the critical feedback frequencies to selectively reduce the gain at only those frequencies. The

drawback to this method is that the feedback frequencies may vary in different acoustical environments, thereby requiring adjustments to the effective frequencies of the filter for each environment (Holube, I. and Velde, T.M. 2000).

In digital hearing instruments, a feedback reduction algorithm reduces or eliminates feedback detected in the input signal. An adaptive filter adjusts within the signal processor to eliminate the feedback path in the specific frequency region. The end result is a highly selective frequency filter that self-adjusts to various acoustical environments (Holube, I. and Velde, T.M. 2000). This avoids the problems associated with feedback encountered with analog hearing instruments.

Expansion

By amplifying all low intensity sounds, hearing instruments inevitably increase the gain of unwanted low intensity sounds such as microphone and circuit noise that are the result of the amplification process. Expansion, an additional feature not available on analog hearing instruments, is a strategy designed to minimize the amplification of very low intensity sounds. With expansion, low intensity inputs (10-20dB SPL) receive less gain in order to reduce the presence of microphone and circuit noise in the output signal (Venema, T.H. 1998). Expansion may be a comfort-enhancing feature if the user is able to perceive the low-level internal noise that is being amplified, however, this comfort may come at the cost of intelligibility (Walker, G., Byrne, D., and Dillon, H. 1984).

CIRCUITRY OPTIONS

Analog and digital hearing instruments may provide linear amplification with peak clipping, linear amplification with compression limiting, or wide dynamic range compression (WDRC).

Linear Peak Clipping

Linear amplification with peak clipping provides constant gain as a function of input level. The decibel increase in the amplified signal is equal to the decibel increase in the input signal until the input signal reaches the saturation point of the amplifier. When the amplifier is saturated, the output signal is limited by peak clipping. As a result of peak clipping, amplitude peaks are removed from the amplified signal resulting in the deletion of some of the original input signal and the introduction of distortion in the output signal. Distortion present in the output signal can have a negative impact on sound quality and intelligibility (Dempsey, J.J. 1997).

Limiters

Compression limiting hearing instruments, or limiters, also provide constant gain as a function of input level. The decibel increase in the amplified signal is equal to the decibel increase in the input signal until the input signal reaches the kneepoint or compression threshold, of the hearing instrument. Input signals with intensities above the kneepoint of the hearing instrument are amplified less than those occurring below the kneepoint as opposed to removing portions of the output signal as in peak clipping instruments (Venema, T.H. 1998). Limiting the output of the hearing instrument in this fashion avoids the waveform distortions associated with peak clipping devices (Dillon, H. 1996), thereby resulting in greater listening comfort.

Wide Dynamic Range Compression

Wide dynamic range compression (WDRC) circuitry provides linear gain to low intensity inputs (less than 40 dB SPL), but begins to compress at lower kneepoints than limiters. For example, a typical limiter has a kneepoint above 65dB SPL whereas a typical WDRC instrument has a kneepoint below 65dB SPL. As a result, WDRC provides a gradual gain reduction over a wide range of inputs (Venema, T.H. 1998) and is generally used for listeners with tolerance problems. The gradual amplification of all but the soft inputs is thought to be more tolerable to this type of listener than peak clipping and limiting circuitry (Dempsey, J.J. 1997).

Circuitry Comparisons

There have been numerous studies comparing the various types of compression and linear circuitry to one another (Hickson, L.M.H. 1994; Dillon, H. 1996). The evidence provided indicated no distinct superiority between linear or a particular compression strategy or between the various types of compression (described in the preceding section) provided the listener is allowed to manipulate the volume control on the linear hearing aid (Killion, M.C. and Fikret-Pasa, S. 1993). The ability to lower the volume control on the linear hearing aid when the input levels are such that it is being driven into saturation reduces the distortion present in the output signal. This manipulation is performed automatically with compression circuitry.

Subjectively, past studies have found that listeners prefer the sound quality and clarity of compression limiting to linear peak clipping circuitry when the two circuits are saturated and volume adjustments are not allowed on the linear hearing aid (Hawkins, D.B. and Naidoo, S.V. 1993). When the listener is allowed to change the

volume control, there is little preferential difference between compression and non-compression hearing aids (Hayes, D.E. and Cormier, K.L. 2000).

ANALOG VS. DIGITAL HEARING INSTRUMENTS

Previous studies comparing analog and digital hearing instruments have shown mixed results which coincide with the design of the respective study (Arlinger, S., Billermark, E., Oberg, M., Lunner, T., Hellgren, J. 1998; Valente, M., Fabry, D.A., Potts, L.G., Sandlin, R.E. 1998; Berninger, E., Karlsson, K.K. 1999; Boymans, M., Dreschler, W.A., Schoneveld, P., Verschuure, H. 1999; and Bille, M., Jensen, A., Kjaerbol, E., Vesterager, V., Sibelle, P., Nielson, H. 1999).

New Digital Hearing Instrument vs. Users' Current Analog Hearing Instrument

Several studies have compared new digital hearing aids to analog hearing aids currently owned and used by the subjects (Arlinger et al 1998; Valente et al 1998). Results of the Arlinger et al (1998) study indicated a small objective advantage and a strong subjective preference for the digital hearing instrument. The Valente et al (1998) trial found no significant objective difference.

Because neither of the above-mentioned studies was blinded, subjective preferences observed in the above-mentioned studies may have been influenced by the lack of blinding. In a review of non-blinded studies, Benson (1996) found that subjects report preferences for experimental or higher technologies when compared to non-experimental or current technologies. In the above-mentioned studies, the subjects were evaluated using their current hearing instrument and a new, experimental hearing instrument with digital technology. Therefore, it is possible that the preference for the digital hearing instruments was a result of non-blinding.

Additionally, the above studies may have been influenced by circuitry differences between the digital test instruments, which utilized compression technology, and the analog reference instruments, which utilized several types of circuitry including linear. Past clinical trials comparing compression instruments to linear instruments indicated a preference for non-linear amplification (Benson, D., Clark, T.M., and Johnson, J.S. 1992; Moore, B.C., Johnson, J.S., Clark, T.M., and Pluvinage, V. 1992; Parving, A., Sorup Sorensen, M., Carver, K., Christensen, B., Sibelle, P., and Vesterager, V. 1997). Thus, the circuitry differences between the test instruments and the reference instruments in the above studies may have affected the subjective results.

<u>New Digital Hearing Instrument vs. New Analog Hearing Instruments</u>

Additional studies have compared newly fit digital hearing aids to newly fit analog hearing aids (Berninger et al 1999; Boymans et al 1999). Results of the Berninger et al (1999) study indicated no significant objective difference, but a subjective preference for the digital hearing aid on the Abbreviated Profile of Hearing Aid Benefit (APHAB) (Cox, R.M. and Alexander, G.C. 1995) in the Aversiveness to Sound (AV) category and in the mean total aided benefit scores reflecting speech recognition which is comprised of the Ease of Communication (EC), Reverberation (RV), and Background Noise (BN) categories.

Data for the Boymans et al (1999) study were collected at two different sites. One site, AMC, consisted of experienced hearing aid users, and for objective testing, used a standard 5-second noise interval to precede the test sentences. The second site, EUR, consisted of new hearing aid users and utilized a longer noise interval to activate the noise-reduction algorithm of the digital hearing aid during the objective portion of

testing. The participants at the AMC site performed significantly worse in continuous noise with the digital hearing aid than with the analog hearing aid, whereas the EUR site participants performed significantly better with the digital hearing aid in continuous noise and in car noise than with the analog hearing aid. Researchers attributed the performance differences between the two groups to differences in experimental protocol. Subjective data were obtained via a questionnaire rating six acoustic situations: in quiet, in a car, on a telephone, watching television or being in a theater, and listening to music. Eleven aspects of acoustic situations were rated significantly better for the digital hearing aid than the analog hearing aid. The total subjective score was also higher for the digital than the analog hearing aid.

As mentioned previously, studies comparing technologies may be affected by a lack of blinding. In the above-mentioned studies, the subjects were fit with new analog reference hearing aids as compared to using analog hearing aids that they already owned. Therefore, a blinded protocol was not followed for the experiments, and this may have affected the results of the studies.

It should also be noted that as mentioned previously, circuitry differences between compression hearing instruments and linear instruments may affect the participants' subjective assessment of the hearing instrument's performance. Neither the Berninger et al (1999) study nor the Boymans et al (1999) study utilized the same processing strategy for both the test and reference hearing aids, therefore, circuitry differences between the test instruments and the reference instruments may have affected the subjective results.

Blinded Study Comparing Digital to Analog Hearing Instruments

One study was found that attempted to compare digital hearing aids to analog hearing aids under blinded conditions. A study by Bille et al (1999) compared a select digital hearing aid to a select model of analog hearing aid under blinded conditions. In this study, no significant objective differences were found between the digital hearing instrument and the analog hearing instrument. Subjectively, no significant differences were found between the digital hearing instrument and the analog hearing instrument regarding overall preference or overall satisfaction. The only significant subjective difference found between the digital hearing instrument and the analog hearing instrument was that subjects indicated that traffic noise was convenient or less annoying when using the digital hearing instrument.

It should be noted that although the above-mentioned experiment was blinded, the analog hearing aid utilized as the reference did not have the same signal processing circuitry as the digital hearing aids tested, and the experimenters utilized different fitting methodologies to fit the hearing aids. The analog hearing aid utilized in this experiment had a conventional user-operated volume control and employed an output limiting compression scheme, which, as mentioned previously, amplifies in a linear fashion until the output of the instrument reaches a pre-determined kneepoint. The digital hearing in this study aid utilized an automatic volume control (changes cannot be affected by the user), and employed wide dynamic range compression (WDRC), which as mentioned previously, provides gradual amplification to all but the soft input sounds. It is possible that analog versus digital effects were present, but the greater gain of the analog hearing aid at the moderately high input level utilized in this experiment (65dB

SPL) overcame the beneficial effects that the digital WDRC instrument offered. Had this experiment controlled for circuitry differences in the blinded condition, objective results and or subjective results may have shown a significant difference between types of hearing instrument.

Additionally, the experimenters utilized different fitting methodologies for the respective hearing aid tested. The National Acoustic Laboratories (NAL) pure-tone threshold-based procedure was used to fit the analog hearing instruments (Byrne D. and Dillon, H. 1986), and the digital instrument was fit using the manufacturer's software and a modified Hughson-Westlake technique (Carhart, R. and Jerger, J.F. 1959). By not controlling the prescriptive method by which the respective type of hearing aid was fit, the experimenters may have introduced additional confounds to the intended purpose of the study.

RATIONALE

Previous studies comparing digital and analog hearing aids have demonstrated no definitive objective advantage favoring digital hearing aids and have shown only mixed subjective preference between the two types. However, none of the previous studies reviewed comparing digital to analog hearing aids controlled for circuitry type, and several of the studies did not use blinding in the design of the study, which may confound subjective studies due to the Hawthorne effect (Valente et al 1998; Bentler, R.A. 2000).

As previously mentioned, past studies comparing digital to analog hearing aids demonstrated questionable if any objective or subjective preference in the favor of digital hearing aids under varying condition of control for blinding or circuitry. It is

possible that the past studies may be inconsistent because the results reflect differences between the test hearing aid and the reference hearing aid in terms of being blinded to the introduction of new technology (digital signal processing). Additionally, differences between analog and digital performance revealed in past studies may have been influenced by circuitry differences between the two hearing aids being compared rather than exclusively determining the benefits of digital signal processing.

The purpose of the present study is to determine if objective and/or subjective differences between analog and digital hearing aids exist when blinding is utilized in the protocol and circuitry is controlled.

CHAPTER III

METHODOLOGY

SUBJECTS

Ten normal hearing and seven hearing-impaired persons participated in this study. Criteria for normal hearing sensitivity was based on a) pure tone air conduction thresholds for each ear between 0 dB HL and 20 dB HL from 250 Hz to 8000 Hz (ANSI S3.6-1996) (Figure 1, Table 1a), b) normal tympanograms bilaterally, and c) unremarkable otoscopy. Criteria for hearing-impaired subjects included a) pure tone air and bone conduction thresholds exceeding 20 dB HL in at least four of the six octave interval frequencies from 250 Hz to 8000Hz (Figure 1, Table 1b), b) normal tympanograms bilaterally, and c) unremarkable otoscopy. All qualification and experimental tests were conducted in a sound-treated examination room (Industrial Acoustic, #105884) with ambient noise levels suitable for testing with ears uncovered (ANSI S3.1-1991).

STIMULI

The Central Institute for the Deaf 50 word lists (CID W-22, list 1A and list 2A) and the Hearing in Noise Test (HINT) (House Ear Institute) served as the stimuli. Speech stimuli and background noise were reproduced by a compact disc player and routed through a two-channel diagnostic audiometer (GSI-61) to a loudspeaker located in the sound treated examination room. The output level of the speech stimuli and background noise were calibrated at the vertex of the listener and were checked periodically throughout the experiment.

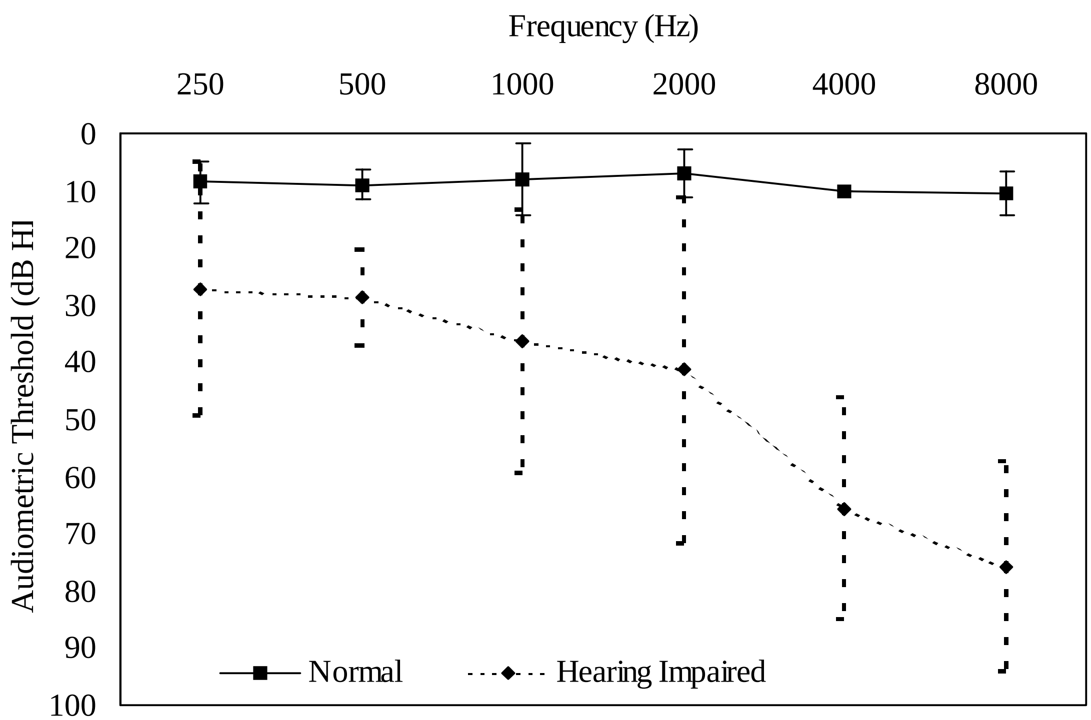


Figure 1. Average audiometric threshold for normal hearing and hearing-impaired subjects.

Table 1a. Audiometric Threshold values for each normal hearing subject.

Frequency in Hz

Subject	250	500	1000	2000	4000	8000
1	0	5	0	5	10	5
2	5	5	0	5	0	5
3	5	5	10	5	15	20
4	10	20	10	10	10	10
5	10	10	10	10	10	10
6	15	15	20	10	10	15
7	15	15	10	5	15	10
8	5	0	0	5	5	10
9	10	5	5	5	15	10
10	10	10	15	10	10	10

Table 1b. Audiometric Threshold Values for each hearing impaired subject.

Frequency in Hz

Subject	250	500	1000	2000	4000	8000
1	30	25	35	35	65	80
2	65	50	50	45	55	65
3	10	20	40	30	60	70
4	10	15	30	50	60	90
5	20	30	30	40	60	70
6	35	40	35	40	70	75
7	20	20	35	50	90	80

HEARING INSTRUMENTS

Qualified subjects were fit unilaterally with an analog Unitron Sound F/X behind-the-ear hearing aid and with a digital Unitron Sound F/X behind-the-ear hearing aid in randomized order. Both hearing instruments utilize wide dynamic range compression circuitry (WDRC), and the noise reduction and expansion features of the digital hearing instrument were disabled during all phases of the study.

EXPERIMENTAL PROCEDURES

Subject Qualification

Qualification testing was performed in the Louisiana State University Amplification Laboratory located in the Music and Dramatic Arts Building. Subjects exhibiting normal hearing sensitivity (pure-tone air conduction thresholds no greater than 20 dB HL), Type A tympanograms, and normal otoscopic findings qualified as normal hearing listeners. Listeners exhibiting at least a mild hearing loss (pure-tone air and bone conduction thresholds greater then 20 dB HL in at least four frequencies tested), Type A tympanograms, and normal otoscopic findings qualified as hearing impaired.

Hearing Instrument Fitting

The analog and digital hearing instruments were programmed for each subject using the subject's audiometric information and the desired sensation level (DSL) fitting strategy (an audiogram reflecting thresholds of 50 dB HL was utilized for each normal hearing subject). No uncomfortable loudness level (UCL) data was measured, therefore, predicted UCL values were utilized for all subjects.

The analog and digital behind-the-ear (BTE) hearing aids were coupled to each subject using a disposable comply earmold with the non-test ear being occluded with an Etymotic Acoustic Research (EAR) disposable earplug. Probe microphone measurements were made on each subject to verify appropriate fit of both hearing instruments using a swept pure tone at two levels (40 and 70 dB SPL).

The probe microphone system measurements consisted of 65 data points measured in $1/12^{th}$ octave steps over a frequency range of 200 Hz to 8000 Hz. Data for output levels at the tympanic membrane in the four conditions stored in the Audioscan RM500 were downloaded to a personal computer using the Audioscan RM500 XDATA32 data extraction program.

Study I

The first purpose of the present study was to evaluate listener performance in quiet when utilizing an analog or digital hearing aid. Word recognition testing was performed in quiet with the subjects seated 1 meter from the loudspeaker located at 0 degree azimuth in the sound treated room. The CID W-22 word list was presented at levels representative of soft speech (40 dB SPL) and loud speech (70 dB SPL) via an audiometer routed to the loudspeaker at 40 and 70 dB SPL for the following conditions:

1) Analog hearing instrument at 40 and 70 dB SPL
2) Digital hearing instrument at 40 and 70 dB SPL

Study II

The second purpose of the present study was to evaluate listener performance in noise when utilizing an analog or digital hearing aid. The subject was seated 1 meter from the loudspeaker located at 0 degree azimuth in the sound treated room. The HINT

was administered at levels representative of soft speech (40 dB SPL) and loud speech (70 dB SPL). The HINT sentence stimuli and background noise was routed through the audiometer to the loudspeaker located 1 meter from the subject at 0 degree azimuth for the following conditions:

1) Analog hearing instrument at 40 and 70 dB SPL
2) Digital hearing instrument at 40 and 70 dB SPL

Test reliability was determined by measuring the HINT score twice for each condition at each sentence presentation level. An average of the two trials served as the mean HINT score for that subject in the given condition. In the event the HINT score for the first and second trial disagreed by greater than 2 dB, a third trial was performed, and the average of the three trials served as the HINT score. HINT scores were analyzed to evaluate the subject's performance in noise when fit with analog and digital hearing instruments.

Study III

The third purpose of this study was to subjectively evaluate hearing aid performance. Hearing aid performance was determined via questionnaire, which was administered following all other testing (See Appendix A).

EXPERIMENTAL PROTOCOL

Prior to testing, the subjects were given verbal and written instructions describing the experiment and their task. It was explained that the purpose of the study was: 1) to determine whether or not subjects perform better in quiet with either an analog or digital hearing aid via word recognition testing, and 2) to determine if subjects perform better in the presence of background noise with analog or digital

hearing instruments, and 3) whether or not the subjects prefer one hearing instrument to the other. The conditions for testing were as follows:

1) Aided with the analog hearing instrument
2) Aided with the digital hearing instrument

Prior to data collection, an experimental schedule was generated for each subject listing a completely randomized assignment for test condition, sentence list, and presentation level. The subjects were blinded as to which instrument they were fit with at all times. The test instruments utilized in this study were identical in appearance. Experimental testing was completed in one session with breaks as needed. The following protocol was used for each subject:

Subject Qualification

1) Sign Consent Form
2) Audiological evaluation

Experimental Procedure

1) Insert probe tube into right ear of subject
2) Conduct probe microphone measurements using a swept pure tone in condition 1 at 40 and 70 dB SPL
3) Conduct word recognition test in quiet in condition 1
4) Conduct the HINT in condition 1 at 40 and 70 dB SPL
5) Repeat steps 2, 3 and 4 in condition 2
6) Administer questionnaire

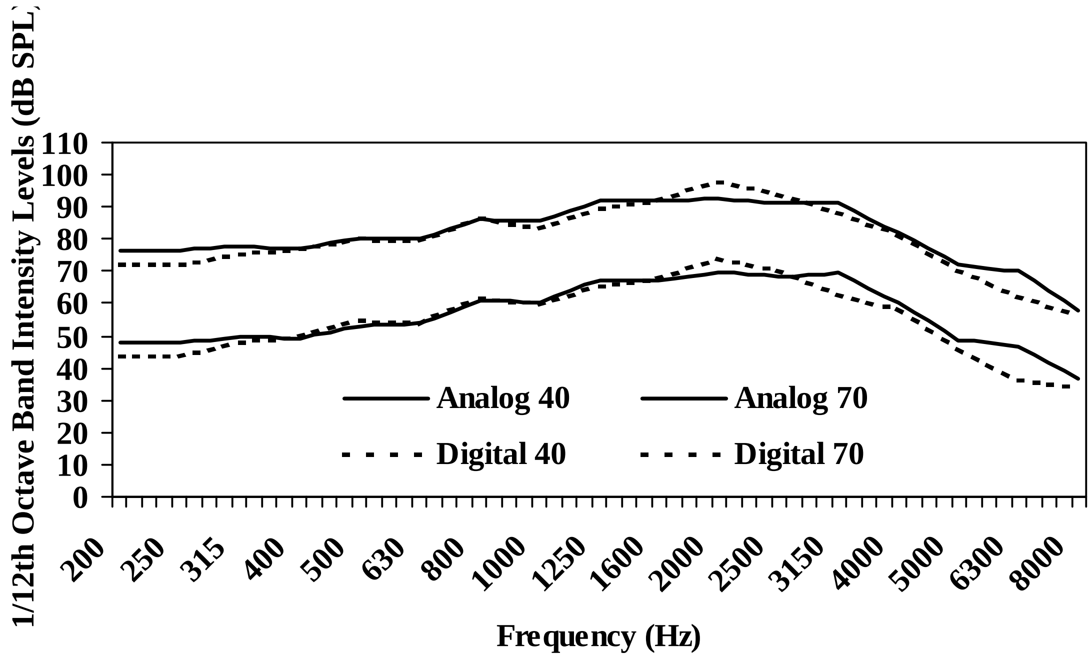
CHAPTER IV

RESULTS

Probe Microphone Measures

Probe microphone measures obtained at the plane of the tympanic membrane were averaged across seventeen subjects for each condition and each intensity level (Figure 2). A mean output level was then calculated for each hearing aid at each intensity level by averaging the output levels at 250, 1000, 2000, 3000, 4000, 6000, and 8000 Hz (Figure 3).

A two-way analysis of variance was performed to evaluate the effects of hearing aid and intensity level. The dependant variable was mean output level. The within-subject factors were hearing aid with two levels (analog and digital) and intensity level with two levels (40 and 70 dB SPL). The between-subject factor was group with two levels (normal and hearing impaired). The analysis revealed significant main effects for level $F(1,15)=5507.451$, $p<0.05$, significant main effects for group $F(1,15)=16.808$, $p<0.05$, as well as a significant level by group interaction $F(1,15)=16.238$, $p<0.05$. No significant main effects were found for hearing aid $F(1,15)=2.608, p>0.05$, aid by group $F(1,15)=0.020, p>0.05$ or aid by level $F(1,15)=0.056, p>0.05$ (Table2). The lack of interaction with the hearing aid variable indicated that the mean output levels from the analog hearing aid were not significantly different from the mean output levels of the digital hearing aid at either level for either group.



<u>Figure 2.</u> Average probe microphone measurements for the analog and digital hearing aids at 40 dB SPL and 70 dB SPL.

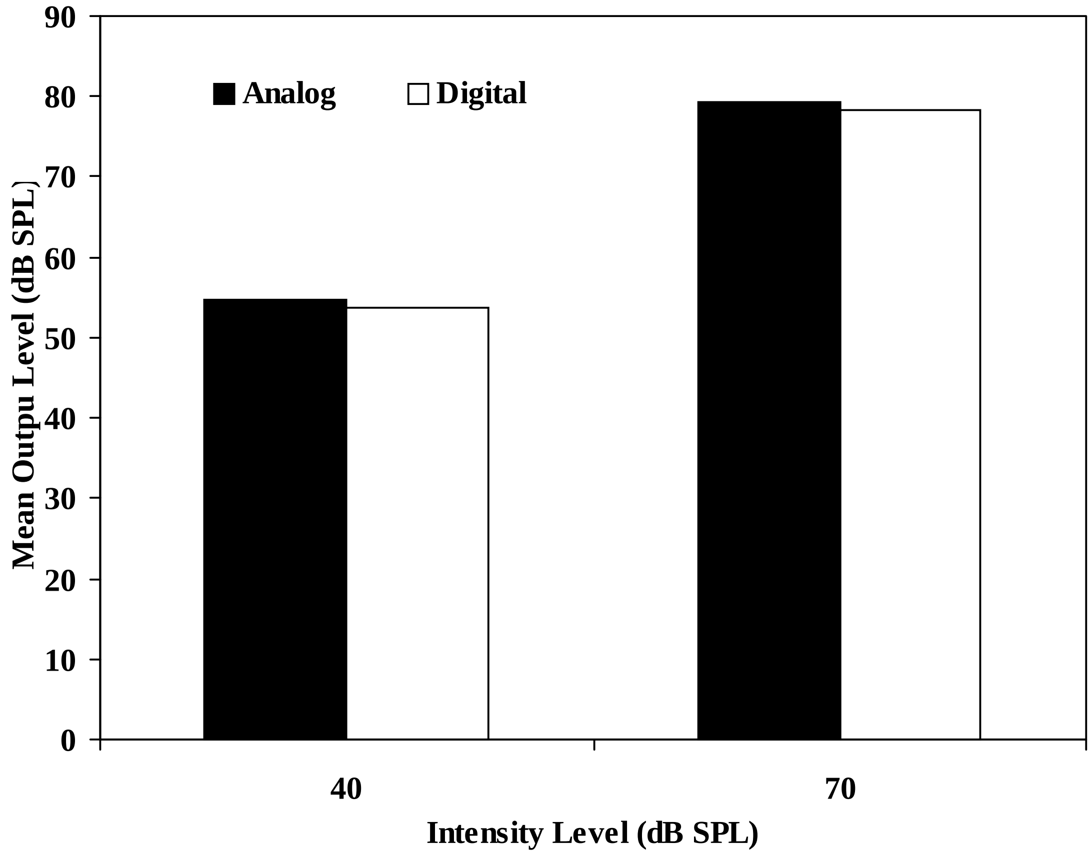


Figure 3. Mean output level at 40 dB SPL and 70 dB SPL for the analog and digital hearing aid.

<u>Table 2</u>. Results of within-subjects analysis of variance for probe microphone measures.

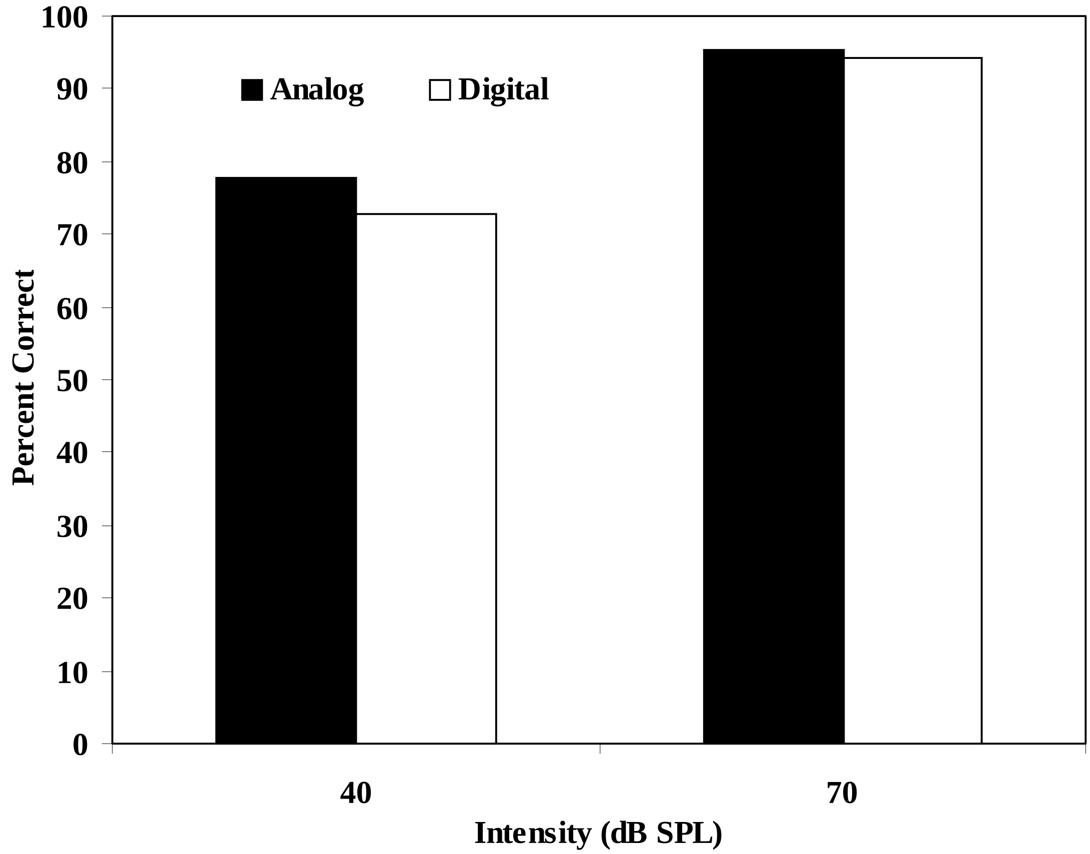
ANOVA	**F**	**dF**	**Significance**
Level	5507.451	1,15	0.000
Level*Group	16.238	1,15	0.001
Hearing Aid	2.608	1,15	0.127
Aid*Group	0.020	1,15	0.890
Aid*Level	0.056	1,15	0.817

Study I

The first purpose of the present study was to evaluate listener performance in quiet when utilizing an analog or digital hearing instrument. Word recognition testing was conducted at two levels (40dB SPL and 70dB SPL) for each subject. Word recognition scores were then averaged across seventeen subjects for each condition and intensity level (Figure 4).

A two-way analysis of variance was performed to evaluate the effects of hearing aid and intensity level. The dependant variable was word recognition score. The within-subject factors were hearing aid with two levels (analog and digital) and intensity level with two levels (40 and 70 dB SPL). The between-subject factor was group with two levels (normal and hearing impaired). The analysis revealed significant main effects for hearing aid $F(1,15)=8.851, p<0.05$, intensity level $F(1,15)=55.594, p<0.05$, and for group $F(1,15)=120.765, p<0.05$ as well as a significant aid by group interaction $F(1,15)=13.039, p<0.05$ (Table 3).

These results indicated that normal hearing listeners performed significantly better in quiet than hearing impaired listeners and that performance was significantly better when presentation levels were increased. These results also indicated that listeners performed significantly better in quiet with the analog hearing aid than with the digital hearing aid.



<u>Figure 4.</u> Average word recognition score at 40 dB SPL and 70 dB SPL for the analog and digital hearing aid.

Table 3. Results of analysis of variance for Word Recognition Scores (Study I).

ANOVA	F	dF	Significance
Hearing Aid	8.851	1,15	0.009
Intensity Level	55.594	1,15	0.000
Group	120.765	1,15	0.000
Aid*Group	13.039	1,15	0.003

Study II

The second purpose of the present study was to evaluate listener performance in noise with analog or digital instruments. The HINT was administered at two levels (40dB SPL and 70dB SPL) for each subject. HINT scores were then averaged across the 17 subjects for each condition and intensity level (Figure 5).

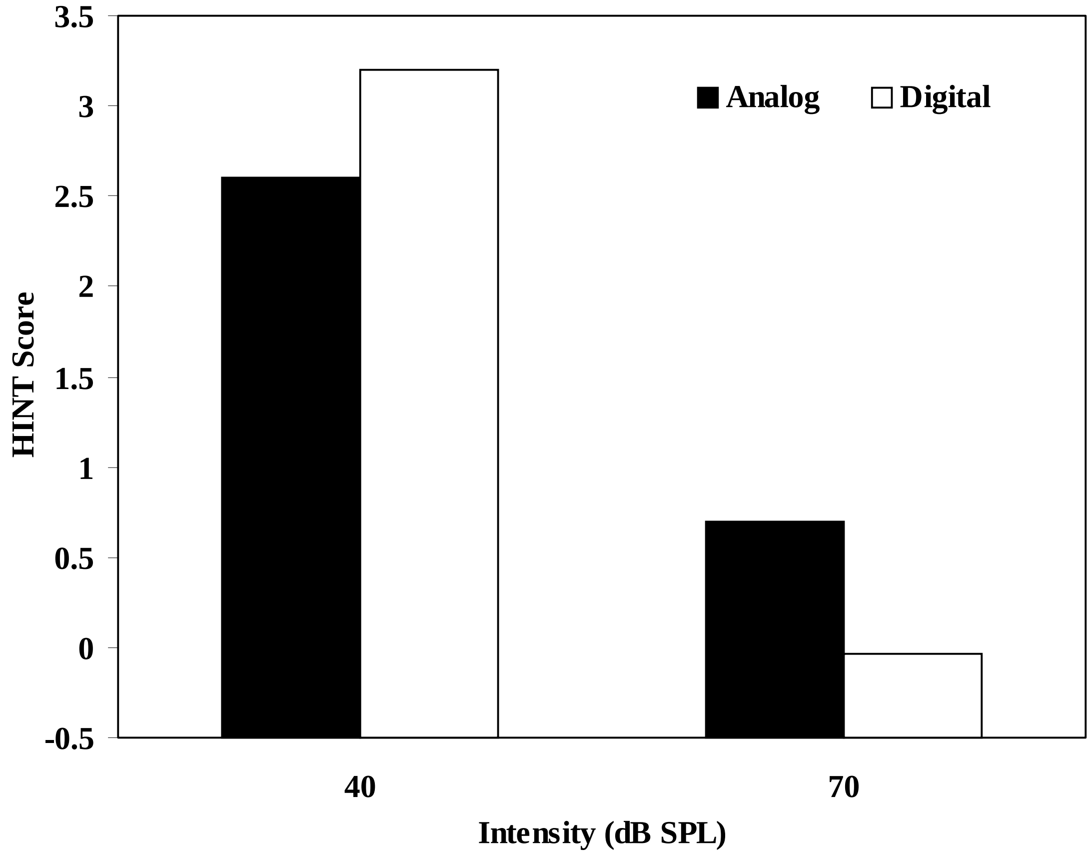
A two-way within-subjects analysis of variance was performed to evaluate the effects of hearing aid and intensity level. The dependant variable was HINT score. The within-subject factors were hearing aid with two levels (analog and digital) and intensity level with two levels (40 and 70 dB SPL). The between-subject factor was group with two levels (normal and hearing impaired). The analysis revealed significant main effects for level $F(1,15)=18.778, p<0.05$ and group $F(1,15)=29.209, p<0.05$ as well as a significant hearing aid by level interaction $F(1,15)=5.192, p<0.05$ (Table 4).

These results indicated that normal hearing listeners performed significantly better in background noise than hearing impaired listeners and that performance was significantly better when the presentation level was increased. However, listener performance in noise was not significantly improved with either hearing aid.

Study III

The third purpose of the present study was to subjectively evaluate hearing aid performance. Each subject was administered a questionnaire following all other testing (Appendix A). The results of subjective testing are listed in Figures 5a, 5b, and 5c.

A one-sample chi-square test was conducted for each questionnaire item to assess listener opinion. The results of the test indicated that there were no significant



<u>Figure 5.</u> Average HINT scores at 40 dB SPL and 70 dB SPL for analog and digital hearing aids.

Table 4. Results of analysis of variance for the HINT (Study II).

ANOVA	F	dF	Significance
Level	18.778	1,15	0.001
Group	29.209	1,15	0.000
Aid*Level	5.912	1,15	0.028

<u>Table 5a.</u> Subjective evaluation of speech reception performance for the analog instrument, digital instrument, or perceived equal performance.

	Observed N	**Expected N**
Analog	5	5.7
Digital	8	5.7
Equal	4	5.7

<u>Table 5b.</u> Subjective evaluation of sound quality for the analog instrument, digital instrument, or perceived equal sound quality.

	Observed N	**Expected N**
Analog	5	5.7
Digital	9	5.7
Equal	3	5.7

<u>Table 5c.</u> Subjective evaluation of overall preference for the analog instrument, digital instrument, or perceived equal preference.

	Observed N	**Expected N**
Analog	5	5.7
Digital	9	5.7
Equal	3	5.7

findings for any of the questionnaire items (Table 5). These results suggested that listeners did not favor one aid type over the other in terms of individual instrument performance, sound quality, or overall preference.

These results did not indicate a significant difference in the subjective evaluation of the hearing aids.

Table 6. Non-Parametric Chi-Square test of subjective test items (Study III).

	Chi-Square	dF	Significance
Item 1	1.529	2	0.465
Item 2	3.294	2	0.193
Item 3	3.294	2	0.193

CHAPTER V

DISCUSSION

Probe Microphone Measures

Probe microphone measures were made to determine the response characteristics of each hearing aid for each condition. The results of the probe microphone measures indicated that the mean output levels from the analog hearing aid were not significantly different from the mean output levels of the digital hearing aid at either level for either group. Stated differently, each hearing instrument had similar output spectra at the tympanic membrane for the two levels tested for each listener. Therefore, any potential objective and/or subjective performance differences between the aids may be attributed to the different means of signal processing rather than differences in circuitry and/or fitting methodologies.

Study I

The first purpose of the present study was to evaluate listener performance in quiet when utilizing an analog or digital hearing instrument. The results of the word recognition in quiet indicated that normal hearing listeners performed significantly better in quiet than hearing impaired listeners and that performance was significantly better when presentation levels were increased. Significant main effects for group and for level were expected due to the fact that normal hearing listeners perform better on word recognition tests than hearing impaired listeners and that performance intensity functions indicate that identification increases as presentation level increases for each group (Penrod, J.P. 1994).

A significant main effect for hearing aid was not expected due to the similarity of the measured output levels of each hearing aid at each level. However, the results indicated that listeners performed significantly better in quiet with the analog hearing aid than with the digital hearing aid. Upon further investigation, it was found that the word recognition scores for the normal hearing group appeared equivalent at each level, however, the word recognition scores of the hearing impaired group demonstrated a decreased percentage of correct responses at the low presentation level with the digital hearing aid. For example, for the normal hearing group, the average word recognition scores were 98.2% with the analog instrument and 98.6% for the digital instrument at the 40dB SPL level. The hearing impaired subjects, however, averaged 48.6% with the analog instrument and 36.0% with the digital instrument. The scores for the hearing impaired subjects were on average 12.6% better at the 40dB SPL input level with the analog hearing instrument.

One possible explanation for the decreased performance with the digital instrument is that the digital hearing instrument may have utilized too low a bit number, causing poor signal fidelity for low-intensity sound reproduction. The number of bits available for signal description determines the resolution of the digital signal.

A bit corresponds to either voltage on, represented by the value of 1, or voltage off, represented by the value of 0. To increase the resolution of the digitized signal, the number of potential values describing the signal must be increased, which is accomplished by increasing the number of bits. For example, a one-bit analog-to-digital (A/D) converter has two potential values to assign to a sample point. A two-bit A/D converter has four possible values and a four-bit A/D converter has 16 possible values.

Thus, using a greater number of bits in A/D conversion results in higher resolution or fidelity of the digital signal (Schweitzer, C. 1998).

Digital systems that use an inadequate number of bits in A/D conversion may result in quantization error, which is the difference between the original analog input signal and the digitized output signal. Quantization error produces a low intensity random noise that determines the noise floor of a digital signal processor. The noise floor, in turn, determines the dynamic range of the digital instrument. Increasing the number of bits in the A/D converter can control quantization error. Each additional bit utilized in a digital system decreases quantization error and thereby decreases the noise floor of the instrument by 6dB SPL (Holube, I. and Velde, T.M. 2000). Therefore, it is possible that the digital hearing instrument utilized an inadequate number of bits in its processor, and low level noise may have been introduced into the output signal to an extent that the hearing impaired subjects scored more poorly at the low presentation level (40 dB SPL) in a quiet setting.

A second possible explanation for the significantly higher hearing-impaired performance with the analog hearing aid at 40 dB SPL in quiet is that the digital hearing instrument may have had an inadequate sampling rate. In a digital processor, the sampling rate of the analog-to-digital converter determines the discrete points in time that the input signal is sampled. Digital systems must sample the input signal at a sufficient rate in order to preserve a faithful representation of the input signal. Sampling the input signal at an insufficient rate results in the presence of energy at the output of the digital system that was not present in the original input signal. Distortions due to insufficient sampling rate, known as aliasing errors, can be controlled by

sampling the input signal at a rate twice that of the highest frequency of the input signal (Schweitzer 1998).

The inclusion of the expansion feature in a quiet setting may serve to reduce low-level noise introduced by quantization error or insufficient sampling rate. Disabling the expansion feature on the digital instrument utilized in this study may have introduced significant enough levels of noise to reduce the percent correct scores of the hearing impaired subjects at the low intensity presentation level.

Study II

The second purpose of the present study was to evaluate listener performance in noise with analog or digital hearing instruments. The results indicated that normal hearing listeners performed significantly better in background noise than hearing impaired listeners and that performance was significantly better when the presentation level was increased. However, listener performance in noise was not significantly improved with either hearing aid. Significant main effects for group and for level were expected due to the fact that normal hearing listeners perform better than hearing impaired listeners on hearing in noise tasks and that performance tends to increase as a function of the increase in intensity (Penrod, J.P. 1994).

The results further indicated a significant hearing aid by level interaction. Upon further investigation, it was found that there were no significant differences in performance for the normal hearing group for either hearing aid. The hearing impaired group, however, performed significantly better in noise with the analog hearing instrument at the 40 dB SPL presentation level and significantly better with digital hearing instrument at the 70 dB SPL presentation level. Though statistically significant,

this finding may not be clinically significant because the difference between the analog and digital signal-to-noise scores was only .6dB SPL for the 40dB SPL input and .66dB SPL for the 70dB SPL input.

The results of this study did not indicate reduced performance in noise at the low-intensity presentation level with the digital hearing instrument as compared to the analog instrument for the hearing impaired group as it had previously in study I in quiet. This may be because the presence of background noise in the test signal had a masking effect on any low level noise that may have been present in the output of the digital hearing instrument, thus negating the possible lower performance with the digital hearing instrument.

If the noise-reduction algorithm in the digital hearing instrument were enabled, the subjects' performance in the presence of background noise when utilizing the digital hearing instrument may have improved.

Study III

The third purpose of the present study was to subjectively evaluate hearing aid performance. The results of the subjective evaluation indicated no significant difference in the subjective evaluation of the hearing aids. Although there initially appeared to be slight overall preference for the digital hearing instrument, further analysis indicated that there were no significant subjective differences between the analog and digital hearing aids. Neither normal nor hearing-impaired subjects reported a significant improvement in performance, sound quality, or general preference based solely on digital signal processing.

The previous studies reviewed indicated a subjective preference for the digital hearing instrument in the unblended studies, whereas the blinded study revealed no significant preference. The subjective findings of the present study were consistent with those of the blinded study previously reviewed.

CONCLUSIONS

In summation, similar performance was indicated for all objective and subjective tasks for both hearing aids with the exception of better performance in quiet at the 40 dB SPL presentation level with the analog hearing aid for the hearing impaired group. These findings indicated fewer objective and subjective differences between analog and digital signal processing than did the comparison studies previously mentioned in which circuitry and/or blinding were not controlled. This study demonstrated that when circuitry is controlled and blinding is utilized, there is no distinct advantage to utilizing a digital processing strategy, and in fact digital processing may be a disadvantage in quiet situations.

Based on the analog and digital hearing aid performance reported by this study, it did not appear that there was a performance difference commensurate with the $878.00 average per instrument price difference between the two types of hearing aids (Strom, K.E. 2001). In this study, with circuitry controlled and under blinded conditions, the digital signal processor did not prevail as the superior product that it is purported to be by hearing aid manufacturers (Ross, M. 2001).

It is possible that the addition of other features available on digital hearing aids that are not available on analog hearing aids, such as expansion and noise reduction, may result in superior performance of the digital product. As reported above, circuit

noise may have caused a reduction in performance for hearing impaired listeners at low presentation levels in quiet. Adding the expansion feature may serve to reduce the presence of that circuit noise thereby increasing performance.

Additionally, instituting the noise reduction feature in the hearing in noise scenario may result in increased performance for the digital instrument with that feature as opposed to an analog hearing aid or a digital hearing aid that does not have noise reduction circuitry. Future research should investigate the effectiveness of expansion and noise reduction circuitry in digital hearing aids.

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APPENDIX A:

QUESTIONNAIRE

1. Do you feel that you performed the speech reception tasks better with one hearing aid as opposed to the other?

   ( ) Hearing Aid #1   ( ) Hearing Aid #2

   ( ) Equal Performance

2. Do you feel that the sound quality of one hearing aid was better than the other?

   ( ) Yes: In what way?

   ( ) Clearness

   ( ) Less Feedback (squealing)

   ( ) Less Distortion

   ( ) Sounded More Lifelike

   ( ) Other: ______________________________

   ____

   Which hearing aid? ( )Hearing Aid #1

   ( )Hearing Aid #2

   ( ) No

3. If it was recommended that you wear a hearing aid and you were given a choice of the two you wore for these tests, which would you choose?

   ( ) Hearing Aid #1   ( ) Hearing Aid #2

   ( ) No Preference

APPENDIX B

SUBJECT CONSENT FORM

Study Title: A Comparative Study of Analog and Digital Hearing Aids

Performance Site: LSU Speech and Hearing Clinic, Music & Dramatic Arts Building, Louisiana State University.

Investigators:

Principal Investigator: Patrick N. Plyler, Ph.D., Assistant Professor
225-388-3934 8:00-5:00

Co-Investigator: Adam B. Lopez, Graduate Audiologist

Purpose of the Study:	The purpose of this study is to compare speech perception ability with a digital hearing instrument as opposed to an analog hearing instrument in quiet and in varying levels of background noise, and to evaluate subject preference between analog and digital hearing instruments.
Subject Inclusion:	Individuals over 18 years of age.
Number of Subjects:	20
Study Procedures:	You will be seated in a sound treated booth in front of a speaker. A hearing aid will be fitted to one ear, while the other will be fitted with a disposable earplug. You will be asked to repeat words in a quiet setting and with varying amounts of background noise. Following the aided testing, you will be asked to fill out a questionnaire regarding your opinion of each hearing aid's performance. Breaks may be taken at any time at your request. Testing will take approximately 90 minutes.
Benefits:	You may not experience any direct benefit from this study, although the data gathered may affect future hearing aid development and clinical dispensing.
Risks/Discomforts:	The procedures to be used follow normal clinical procedures that impose no known risks to participants.
Right to Refuse:	Participation in this study is voluntary. The participant may change his/her mind and withdraw from the study at any time without penalty.

Privacy:	The LSU Institutional Review Board (which oversees university research with human subjects) may inspect and/or copy the study records. Results of the study may be published, but no identifying information will be included in the publication. Other than as set forth above, subject identity will remain confidential unless disclosure is legally compelled.
Financial Information:	There is no cost to the subject, and subjects will not receive financial compensation for participation.

Signatures:

The study has been discussed with me and all my questions have been answered. I may direct additional questions regarding study specifics to the investigators. If I have questions about the subjects' rights or other concerns, I can contact Charles E. Graham, Institutional Review Board, (225) 388-8692. I agree to participate in the study described above and acknowledge the investigator's obligation to provide me with a signed copy of the consent form.

Subject Signature:____________________ Date:_______

Witness Signature:____________________ Date:_______

The study subject has indicated to me that he/she is unable to read. I certify that I have read this consent form to the subject and explained that by completing the signature line above, the subject has agreed to participate.

Signature of Reader:____________________ Date:_______

VITA

Adam Benjamin Lopez is a Louisiana native, born in New Orleans on May 4, 1972. He received a Bachelor of Arts in Communication Sciences and Disorders from Louisiana State University in December 1998. He began his clinical fellowship year in audiology while finishing his Master of Arts program at Louisiana State University. Upon completion of his master's degree and following the completion of his clinical fellowship year in Baton Rouge, Louisiana, he intends to seek employment out of state.