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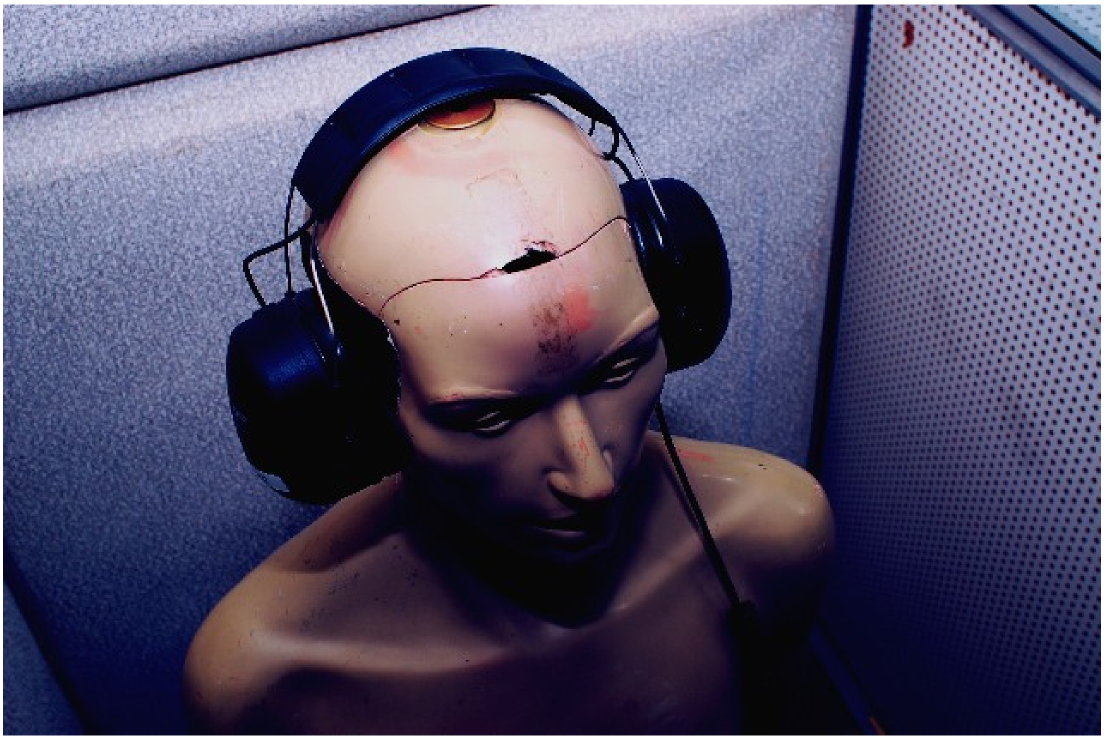

A Speech Intelligibility Testing System

For Evaluation of Communication- and Hearing Protection Devices

Master of Science Thesis in the Master Degree Programme, Sound and Vibration

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Göteborg, Sweden, 2010

Master's Thesis. 2010:152

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Abstract

A testing system for speech intelligibility in a noisy environment was developed and as a first case for testing, hearing protectors with one-way communication possibilities were chosen. The use of such devices leads to the question of the signal exposure needed for successful communications. By the use of Just-Follow-Conversation methods, speech-Bekesy and an Ascending method, a value of signal exposure required for signal intelligibility was determined for a population of 12 subjects. This value was then compared to values acquired from a Hearing In Noise Test for the same product. Using standardized H, M and L filtered pink noise the test was conducted using speech signals in form of audiobooks to minimize the effect of learning. Comparisons of results using a statistical pairwise T-test suggest that the three methods make successful distinction between H-type noise but the methods do not succeed in distinguishing between M- and L-noise, which calls for further investigations into the matter.

Keywords: Speech, Intelligibility, Noise, speech-Bekesy, Ascending, Just-Follow-Conversation, JFC, Hearing in noise test, HINT, Hearing protectors, Communication.

Contents

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I would like to thank those who helped me during the time I worked on this thesis. No names will be mentioned so none are forgotten. You know who you are. I would also like to thank professor Mendel Kleiner for his supervision, although he is the busiest man I know his help was never but an email away and always spot on and much appreciated.

I would also like to thank my girlfriend Elin for her support and encouragements during the 6 months I worked on the thesis, your help was invaluable and I owe you a big one.

Last but not least, I would like to dedicate this thesis to our daughter Ingibjörg, who was born 39 hours and 51 minutes after I presented my thesis, thank you for waiting-That was a close one!

-Ragnar Vidarsson, Göteborg, Sweden

1. Introduction

> We arrive at the knowledge and certainty of some unknown thing only through the knowledge and certainty of another thing that is prior to it in certainty and knowledge.
>
> - B. Spinoza

The importance of intelligibility of messages in communications can not be over emphasized. Reliable flow of information in critical situations is necessary, even imperative, for correct judgment of the information and the decision of a suitable reaction to that information. In noisy environments where hearing protectors are worn the problem of getting information to the wearer can be remedied by the use of loudspeakers inside the cup of the protectors.

At a certain frequency, which is determined by the mass and stiffness of the headband of the hearing protectors, the hearing protectors become transparent to noise. When this happens the signal from the loudspeakers has to compete with external unwanted noise, as well as possible leakage of noise past the protectors. Apart from this issue, hearing protectors never provide perfect isolation to noise; therefore judging the quality of the loudspeakers with regard to intelligibility in noisy situations has to be done by objective or subjective methods.

There is nothing to stop a user from turning up the signal volume to ensure that the signal is raised over the noise in the environment to gain the necessary intelligibility. However, this is not the intended use of hearing protection products, the signal from the loudspeaker can in effect become the problem. Loud noise often degrades a persons hearing, causing total or partial short or long term deafness. This has been established by many, see for example [1, 2] for further discussions on the subject.

Depending on the type of application, different methods are utilized for prediction of intelligibility. Many objective testing systems have been developed, such as the Speech Transmission Index (STI) and a derivative of that, the Room Acoustic Speech Transmission Index (RASTI), the Speech Interference Level (SIL), the Speech Intelligibility Index (SII formerly known as AI)[3], the common denominator for many objective methods is that they are sometimes considered as being insufficient in some aspect, see for example[4, 5, 6, 7, 8].

One reason that is mentioned for this is that the testing methods do not, in general, provide a value that can be understood from a physical point of view, nor do they provide information on what situations they are to be used in so that infallible results are acquired.[4] This often leads to methods being used in situations where they do not apply and thus, give results that do not reflect the situation at hand. In her book, Jekische[4] mentions that this not only applies to laymen but experts as well. Other aspects that are mentioned is the lack of sensitivity to human factors that play a role in intelligibility and a fine discussion of the subject can be found in the book by Jekische.

The need to establish the threshold signal level for intelligibility, three different methods will be used. The methods are not expected to give the same value due to the nature of the test i.e. the method of feedback the listener gives is such that the tests should give independent results. This is done to see if the methods can be used to reinforce each others predictions, and to see which of these methods make independent predictions of the signal levels needed for intelligibility, thus possibly giving a set of methods that one can use to make these predictions. The methods used are known in the field of audiology and are well established [9, 10, 11]. These methods are known as the Hearing In Noise Test(HINT) and two methods of Just-Follow-Conversation(JFC); one that relies upon the Bekesy test and another who is an Ascending method.

The methods are not used in general when classifying communication devices but to establish an absolute threshold of hearing. They may thus be used to provide a view on where a person perceives the signal to be intelligible judging by some criteria even though this is above the threshold of hearing. The possibility of measuring the exposure needed for intelligible speech is also a reason that these tests are chosen. Apart from this, the tests do not apply to any particular situation and are commonly changed to fit the situation at hand[9].

The JFC tests will be implemented in LabView code and post-processing of measurements results will be conducted using MatLab. The thesis is carried out in cooperation with a company well known for its hearing protection products for many years.

1.1. Objective of Thesis

A measure of speech intelligibility, based on signal exposure from a given device, was to be developed. The goal was that this system could be used to test the quality between embedded communication systems in hearing protection devices. This test should fulfill the following criteria

1. The test should be as independent as possible from learning as the group of listeners has to be reusable.
2. The test should be, to some extent, automated so that it does not require more than one person to operate.
3. The test should be usable with different types of noise.
4. The test should not take too long.
5. The LabView program developed has no requirements on the graphical user interface, only on usability.

1.2. Limitations

Although not explicitly stated, the hearing protectors do not contain any electronics apart from the loudspeakers. This means that any form of active control is beyond the scope of this thesis. This does not, however, exclude those types of hearing protectors to be tested. It might in fact be possible to test the effect the noise canceling has on the intelligibility of the signal.

Since the goals state that the focus of the LabView part is on useability, the GUI will be kept as simple as possible. Also, regarding the usability, no doubt the level of complexity will be low, as minimal efforts are placed on the programing part of this thesis. It is not a subject of this thesis to create a professional LabView program but a usable one that serves as a basis for (possible) further development.

Detailed statistical discussions will not be the subject of the thesis. Standard methods will be employed for analysis of data wherever applicable and MatLabs function library will be used for the analysis of these matters.

1.3. Structure of Thesis

Following the introduction, chapter 2 gives the methodology that was applied in conducting this thesis, development of scripts and programs, measurements performed and post processing. Chapter 3 presents the theory that the work is based on, background information on the Audiology tests with references for further reading, a discussion and some theory for the STI method as well as references and some statistical theory that was applied in the post processing. Chapter 4 will present the results from the evaluation of the test system, should it be a success or fail to meet it's intended goals as well as the results from the statistical analysis. A discussion on the results is included in chapter 5. Further discussions on the aspects that could have been researched more thoroughly, elements that could have been handled differently and suggestions for further research are included in chapters 6 and 7.

2. Method

After studying the field of Audiology, the basic tools used in the field, the tests used for hearing measurements and techniques used for diagnosis, three tests were selected. Following this phase, scripts in Matlab started to form and in July 2010, the LabView part of the project was developed with the aid of professional LabView developers who's names shall remain anonymous. The development phase consisted mainly of constructing several VI's that incorporate the necessary functions of the audiology tests, as well as VI's for saving data. Said VI's are developed to reduce the size of the programs and enable the reuse of some parts in other programs.

The MatLab scripts developed have the main function to translate results from the LabView program into appropriate units (e.g. volts and dBA), calculate exposure values for the loudspeakers and statistical analysis.

A measurement of the STI for the loudspeakers was conducted to establish that the loudspeakers indeed are suitable for transmission of signals. This was done using the standard setup available with MLSSA measurement system on a KEMAR torso with rubber ears. For practical reasons, the testing took place without the use of any generated noise- It was impossible to create the high noise levels that were needed to degrade the signal at the site of measurement. The signal used was a 1k Hz sinusoid with an amplitude 1V, and microphones used for the measurements were calibrated according to the requirements stated in the manual for MLSSA[20].

Preparing for measurements, the Matlab files were calibrated so that the same sound pressure levels that were measured and calculated were obtained. This calibration took place at the company's sound lab using a 250 Hz -3 dB sine wave on an artificial ear. The transformation of the relative 0-100 scale in the computer used, to a scale of dBA was done by first measuring the voltage output of the sound device, interpolating the values and fitting them to the relative scale in the computer translating the scale to volts. Data for the headphones used in terms of SPL/V was available so the translation to dBA was simple.

2.0.1. Measurements

Performing each test, a scenario was set up as seen in table 2.1. Each of the test methods was performed with three different noise types, each with their own speech file. From this test, an average signal exposure level was calculated. This also enabled an average signal-to-noise ratio to be found as the noise levels were kept constant. Measurements were conducted at the company on the 22. and 23. of

Table 2.1.: Speech Signal $i = 1, 2, 3, \ldots, k$

n	N_H	N_M	N_L
1	S_i/N_H	S_i/N_M	S_i/N_L
2	S_i/N_H	$\vdots$	S_i/N_L
3	S_i/N_H	$\vdots$	S_i/N_L
$\vdots$	$\vdots$	$\vdots$	$\vdots$
N	S_i/N_H	S_i/N_M	S_i/N_L

September 2010. The participants were all employees of the company, 12 Swedish males, ages 27 to 43 with documentations of good hearing- that is to say, no severe loss of hearing other than age related. The participants were tested in a high volume chamber at the at the company's sound laboratory.

The subjects were instructed to place the hearing protectors on their head and position them so that they were comfortable and that they were sure that minimal leakage of sound would occur. The test subjects received oral instructions on how to conduct the test. The subjects were required to aim for an understanding of half of the spoken words heard (hearing does not imply understanding, hearing only implies that a signal is perceived). When the subjects thought that this had been acquired, feedback was given in form of a button push. The signal level was then changed differently depending on which method was used.

In order to estimate the potential noise leakage around the hearing protectors, the hair growth of each subject was documented. More hair growth should introduce more leakage of noise into the system as the cups are not pressed completely flush against the skin. Each subject was given instructions on seating position in the chamber. The noise was introduced to the environment and a signal was loaded into the correct version of the LabView programs developed and played through the loudspeakers mounted in the hearing protectors.

2.0.2. Paired T-test in MatLab

Upon collecting the measurement results, the data was processed and analyzed using Matlab scripts. One subject had data missing from the HINT test- This data was point estimated so that a full set of measurements(containing all methods with all noise) could be used for the analysis. This was specifically done to enable the T-test to be performed.

It should be noted that the functions used require that the statistics toolbox is available in MatLab. In performing a T-test in MatLab one uses the function

```
[H,P,CI]=ttest(X,Y)
```

where X and Y are the populations to be tested. This gives:

- $CI_{100*(1-\alpha)\%}$= Confidence interval for the true mean for a paired test.
- P= Probability of observing a given result, or one more extreme, by chance should H_0 fail to be rejected. Small values of P cast doubt on the validity of H_0.
- H= Fail to Reject(0) or Reject(1)

To test the normality of the distribution, a Lilliefors test is recommended as the population is small[23] and H_0 does not specify *which* normal distribution to test for.

```
H=lillietest(X)
```

where H_0:normal, is the hypothesis and it follows the same convention as the T-test. Failure of rejection may indicate that the data is normally distributed or that there is just a lack of strong evidence against H_0.[23]

3. Background and theory

As of yet, there is not a single standard way of measuring intelligibility of speech. The lack of SI-units of intelligibility has therefore resulted in a myriad of techniques and tools for the measurement of intelligibility both objective and subjective. In this chapter some theory behind subjective audiology tests, objective measures, considerations on speech material and noise as well as some statistics will be presented.

3.1. Audiology tests

For measurement of hearing acuteness and loss of hearing, different methods within audiology are used. The most common applications are the testing of speakers, listeners and a communication link in the ear.

A disadvantage of audiology tests that has to be taken into account, is that there is no frequency specific information to be found. This can be regarded as a problem since the cochlea is highly frequency selective.[9]

3.1.1. JFC tests

Natural conversation is indeed the most normal presentation of speech for recognition[10]. Methods known as Just-Follow-Conversation (JFC) are methods that obtain this naturalness, also referred to as *the face validity* of a signal. This is done at the expense of loosing control over the presented material, since you cannot say that it is phonetically balanced and such. Judging relatively similar communication systems a test should have a signal that represents common speech to a degree as high as possible.[10] The tests used will be a variation of the Bekesy test referred to as speech-Bekesy and a method based on an ascending signal which will be reffered to as JFC$_A$.

The speech-Bekesy(BEK) test

The Bekesy test is a well established test in audiology. The patient is presented with a pure tone signal, which can be continuous or interrupted, at a certain sound pressure level. The test subject presses a button and the level decreases until the instant when the signal is no longer audible, then the button is released and the sound level increases. The patient then presses the button again until non-audible levels are reached. The test traces monaural thresholds for pure tones. The idea of using this method to home-in on a running speech signal will be utilized to find a threshold of exposure that results in intelligibility of speech.

JFC$_A$

The ascending method applied here is sometimes referred to as a method of adjustment. This means that the user controls the level of stimulus by himself setting the volume at a level he can perceive, or according to instructions. In this implementation, the program starts at an audible level and the user signals this by a push of a button, the level then drops by a random number and starts to increase(ascend), thus taking the user adjustment out of the equation, until audible levels are reached again and the user pushes the button again signaling that he understands what is being said according to instructions.

3.1.2. HINT

A test known as Hearing-In-Noise-Test (HINT) was developed by Nilsson, Soli and Sullivan. This test relies upon so called Speech Reception Threshold in Noise(SRTN) measure. The speech material is in the form of spoken sentences that are phonetically balanced. Test subjects are scored based on the number of correctly repeated words in a sentence and the control person adjusts the level accordingly, further details are found in [11]. The test thus requires a test-subject and a test-controller.

3.2. Speech material and it's difficulty

Speech material varies in its level of difficulty depending on the contents. The many different audio metric tests utilize, in general, five to six different forms of material which are based on phonetically balanced words, p-b sentences, single or multiple syllable words, nonsense words or harmonic signals of some sorts.[9] These methods and their associated perceived difficulty are summarized in table 3.1. Nonsense

Table 3.1.: Speech material and it's anticipated difficulty level.

Material	Difficulty level(easy=1, hard=5)
Nonsense words	5 (hardest)
Single syllable words	4
Two syllable words	3
Full sentences	2
Running speech and numbers	1 (easiest)

words are regarded as most difficult since they have no meaning to the listener. The more the material becomes like running speech, the easier it is understood. This is because of redundancy, or glimpsing[13], of speech, that is to say, people can guess what the contents of a sentence will be depending on other words that subjects may hear. In ISO standard TR4870 the use of running speech is not recommended in general because of non-acoustical effects that play a part in the intelligibility and that one looses control over the long time averages of the speech as well as the structure of the sentences. It is however stated that the use of running speech can be used in some situations and it has been established before, that when testing communication devices a signal should resemble speech as closely as possible [10], [14].

3.3. Noise, different types and their effect on signal presentation

Speech in noise has a lower intelligibility even at small sensoneural hearing damages. It is mainly in the mid frequency range people experience problems, which coincides with the frequency range speech relies upon (500-4k Hz).

When testing in noisy environments, Beranek discusses that a difference in a given system is more easily found if the scores from the intelligibility tests are made to be closer to 50% rather then 100%.[15] The need for articulation to be tested in noise is important because articulation scores may be too high for a reliable prediction to be made if not tested in noise.

Masking of speech because of noise takes place upwards in frequency. This means that a low frequent sound masks other sounds of a higher frequency content. The noise spectrum will thus influence hearing differently depending on its frequency content- More masking is expected of low frequent and mid frequent noise than of high frequent noise. It has been established that a spoken sentence retains its intelligibility independent of the noise[16], which is important because it enables the use of many types of noise using the same speech material. Even more prominent masking occurs when noise frequencies are close to the frequencies of the speech sound. This has it's roots in the function of the Basilar membrane of the inner ear, where two frequency sensitive areas are excited and they happen to overlap they interfere with each other sending a signal to the hearing nerve that the brain has trouble deciphering.[2]

Speech material should be presented with a noise signal with an equal longtime average. If the spectra is different, different S/N ratios will be found for different frequency regions which may be suboptimal. The noise used for the measurements is pink noise, filtered to H, M and L weighted noise according to standard BS EN352-4:2001[17], see also[18]. Information and examples of calculations of these noise spectra can also be found in the aforementioned standard.

3.4. Objective measures of intelligibility

The Speech Transmission Index[19] can be used to test electronic communication systems and a STI measurement of the headphones was carried out using MLSSA 10WI[20]. When using MLSSA to perform the STI measurements, articulation loss of consonants, AL_{cons}, is also presented. This measure will thus also be introduced here.

STI

The determination of STI values is based on measuring the reduction of the signal modulation between the location of the sound source with octave frequencies between 125-8k Hz. The useful signal S, is put related with and compared to the interfering signal N. A modulation reduction factor, m(f), is used to characterize the interference with intelligibility.

$$m(f) = \frac{1}{\sqrt{1 + (2\pi f \cdot \frac{RT}{13.8})^2}} \frac{1}{\left(1 + 10^{-\left(\frac{S/N}{10dB}\right)}\right)} \tag{3.1}$$

here

f is modulation frequency in Hz,
RT is the reverberation time in sec,
S/N is the Signal-to-Noise ratio in dB.

Note that when the reverberation time becomes very small or zero equation 3.1 reduces to

$$m(f) \approx \frac{1}{\left(1 + 10^{-\left(\frac{S/N}{10dB}\right)}\right)} \tag{3.2}$$

That is to say, the modulation of the signal is only dependent on the S/N ratio for very small rooms (with respect to wavelength). The effective S/N is then calculated by

$$X_i = \log_{10} \frac{m_i}{1 - m_i} \tag{3.3}$$

where $i = 1, \ldots, 98$ [21] and then the STI value is given by

$$STI = \frac{X + 15}{30} \tag{3.4}$$

anything between 0.5 to 1 is good to excellent.

Articulation Loss of Consonants

The articulation loss of spoken consonants was established to be important when evaluating speech intelligibility in rooms by Peutz and Klein in 1971 [21] and this measure is presented along with the STI results from MLSSA 10WI. The equation for this is given as

$$Al_{cons} \approx 0.652 \left(\frac{r_{QH}}{r_H}\right)^2 \cdot RT \quad [\%]. \tag{3.5}$$

A good measure is, as one can expect, little loss of consonants. This is taken to be everything less then 11%, with levels below 3% considered to be ideal.

3.5. Small sample statistical tests

The size of the population in a data set dictates whether a z-test or a T-test is to be used. Since in our case, the goals state that the population available for the test is limited, a pairwise T-test will be employed. The T-test is a test used when two normally distributed populations are to be compared, variance of the population is equal or unequal and unknown, and the population is small. It should be noted that the larger a population, the stronger the test will be. A prerequisite of the pairwise T-test is that there is a one-to-one relationship between each element of a population.

Setting up a hypothesis, equality between populations with regard to signal mean required for acceptable intelligibility will be tested.

$$H_0 : \mu_1 = \mu_2$$
$$H_A : \mu_1 \neq \mu_2.$$

The test statistic, when the populations are equal, is then calculated as

$$T = \sqrt{n}\frac{\overline{x} - \overline{y}}{\sqrt{s_x^2 + s_y^2}} \tag{3.6}$$

Note that these equations assume that the sample sizes are equal, a value of one sample corresponds exactly to a value another sample. If this is not the case, other measures must be taken(see for example [22]p.1065). The variance is calculated using

$$\hat{\sigma}^2 = S^2 = \frac{1}{n-1}\sum_{j=1}^{n}(X_i - \overline{X})^2. \tag{3.7}$$

If the sample sizes are not equal an estimate of the degrees of freedom (DoF's) must be made. The DoF's allow the test to detect significance in the population means. The sensitivity is lesser as the DoF's are fewer, which increases the risk of making a false judgment of the results, known as type-1 or type-2 errors.

Choosing an appropriate confidence level α and consulting tables for the t-distribution one can find the critical value, c, and from

$$P(T < c_1) = \alpha \tag{3.8}$$

one can find whether to *reject* or *fail to reject* the hypothesis. If $T > c$ then they hypothesis has failed to be rejected, otherwise it is rejected.

4. Results

Results are presented in the same order as the terms are mentioned in chapter 3.

4.1. Speech material

The speech material that acted as test signals can be seen in table 4.1. Files T02,T04 , T08, T12 come from disc two of the book ”Geniet”[24], the numbers correspond to the tracks of the CD. Files T13, T14, T15 and T16 come from cd 8 of the book ”Hannibal: upptakten”[25]. The files were chosen randomly and the chapters are not consecutive on the CD’s regardless of the numbering, with a few extra options in case they were needed.

Table 4.1.: Speech files and long time average data.

File	Wav_{rms} [dB rel. 1V]	Tested with Noise
T02	-17,3	H
T04	-18,4	M
T08	-17,8	L
T12	-18,1	H
T13	-14,4	M
T14	-18,0	L
T15	-13,8	Backup
T16	-17,4	Backup

4.2. Noise

The noise used is bandpass filtered pink noise according to the standard BS EN352-4[17] and shall be named henceforth H, M and L as they are named in the standard. Information on the long time averages and noise levels can be found in table 4.2. For information on the filtering one should consult the standard.

Table 4.2.: Noise files and important data

Noise type	Long time average [dB rel. 1V]	Noise level [dB rel. 20μPa]
H	-16,8	86.15
M	-17,8	81.65
L	-14,0	80.93

4.3. STI measurements

Measurements conducted with MLSSA10WI resulted in STI and AL_{cons} data for the loudspeakers in the hearing protectors. Results can be seen in figure 4.1 along with the frequency response of the loudspeakers in the region where the STI is calculated, measured with MLSSA10WI in figure 4.2.

MTF Matrix (Calibrated)

Frequency-Hz	125	250	500	1000	2000	4000	8000
level dB-SPL	58.4	65.0	70.4	71.1	59.2	78.4	87.9
m-correction	1.000	1.000	1.000	1.000	0.996	1.000	1.000
0.63	0.984	0.995	0.997	0.997	0.944	0.999	1.000
0.80	0.982	0.994	0.996	0.997	0.943	0.998	1.000
1.00	0.979	0.993	0.995	0.996	0.942	0.998	0.999
1.25	0.976	0.991	0.994	0.996	0.942	0.998	0.999
1.60	0.973	0.990	0.993	0.995	0.942	0.998	0.999
2.00	0.968	0.987	0.992	0.994	0.943	0.998	0.999
2.50	0.961	0.985	0.990	0.993	0.941	0.998	0.999
3.15	0.954	0.981	0.988	0.992	0.941	0.997	0.999
4.00	0.945	0.976	0.985	0.990	0.941	0.997	0.999
5.00	0.935	0.970	0.981	0.988	0.937	0.996	0.999
6.30	0.921	0.963	0.976	0.985	0.936	0.996	0.999
8.00	0.905	0.953	0.970	0.982	0.935	0.995	0.999
10.00	0.882	0.943	0.963	0.978	0.934	0.994	0.998
12.50	0.853	0.929	0.954	0.972	0.931	0.992	0.998
octave MTI	0.923	0.977	0.994	1.000	0.897	1.000	1.000

STI value= 0.967 (0.963 modified)   ALcons= 0.9%   Rating= EXCELLENT

htr230f   MLSSA: STI

Figure 4.1.: Measurements of STI for device.

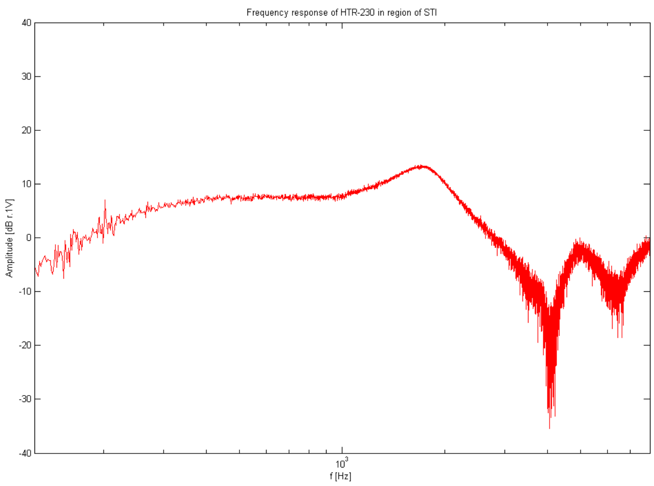

Figure 4.2.: Frequency response in STI region for device used in measurements. Measured on KEMAR doll with calibrated microphones.

4.4. Results from LabView programs and statistical tests

The results from the Labview program were gathered for each test subject and analyzed with Matlab. In this aspect the programs developed performed well. Results from the analysis of data give indications of the exposure from the loudspeaker needed for good intelligibility. First, some descriptive statistics of the signal exposure data acquired from the data set.[1]

Table 4.3.: Descriptive statistics of the signal values acquired after the results from each subject had been analyzed. The values are all presented in dBA.

Variable	N	Mean (μ)	St.dev(σ)	Max	Min
$S_{Bek,H}$	12	37.05	3.15	39.85	32.02
$S_{Bek,M}$	12	44.83	2.87	48.53	38.07
$S_{Bek,L}$	12	44.95	3.32	49.69	37.53
$S_{JFC,H}$	12	40.65	5.07	48.85	33.61
$S_{JFC,M}$	12	45.11	4.53	49.75	32.12
$S_{JFC,L}$	12	45.96	3.06	49.79	40.30
$S_{HINT,H}$	12	44.00	3.15	47.47	41.57
$S_{HINT,M}$	12	47.38	5.07	50.77	45.17
$S_{HINT,L}$	12	47.76	1.97	50.97	46.27

[1]One subject in $S_{HINT,H}$ was point-estimated as data was missing.

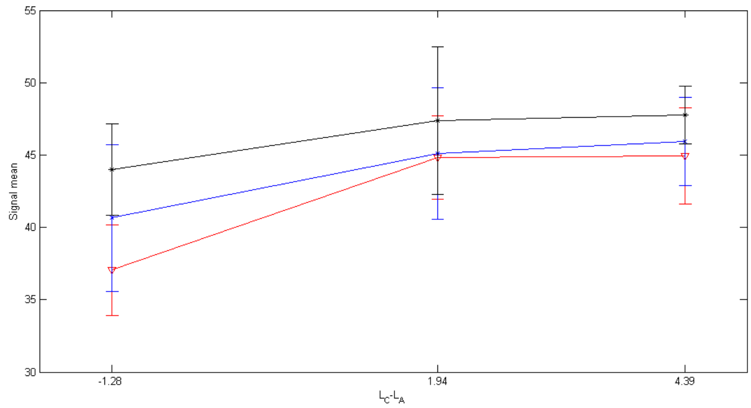

Figure 4.3.: Figure showing the trend of mean signal values with the standard deviation attained for the noise types. Red is BEK, blue is JFC_A, black is HINT.

In figure 4.3, the trend of the development of the signal strength is shown. It is quite clear that when the noise has a lower frequency content, the signal exposure needed for intelligible speech levels increases.

The data in table 4.3 is represented as box plots in figure 4.4. Twelve subjects had complete data across all tests and all noise types, data for these 12 subjects is presented in figure 4.4 and figure 4.5.

The box plot edges extend to the 25th and the 75th percentiles and the central mark is the median of the data set. Percentiles tell us which fraction of the measurements is found below the the limit they present[26]. The error-bounds extend to the most extreme values of the data set not considered to be outliers. Outliers are marked with a red plus mark.

The histogram in figures 4.6, 4.7 and 4.8 give an idea of the normality distribution of the results for each test.

Figures 4.9, 4.10 and 4.11 present a normal probability plot of the data acquired in each method. These plots give further ideas on how the data is distributed.

A normal distribution of the test data is important for the T-test. However, in a paired T-test which tests the differences between two data sets for one subject, the difference between two populations will have a better normal distribution than a single population, this is shown in figures 4.12, 4.13, 4.14, 4.15.

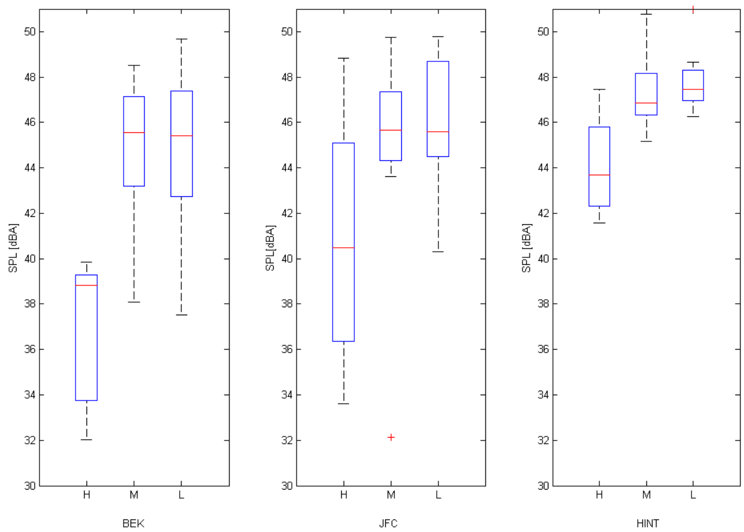

Figure 4.4.: Figure showing a boxplot of how the different noise types perform within each method.

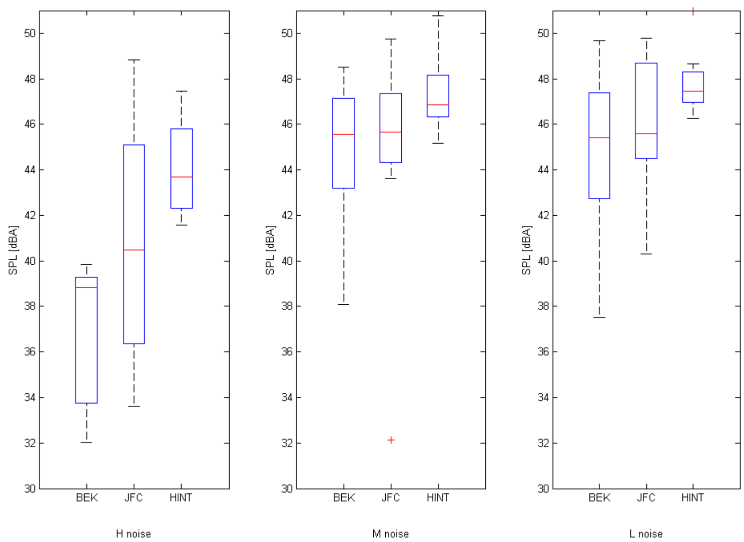

Figure 4.5.: Figure showing how the noise types compare against each other between the Bekesy, JFC and HINT test.

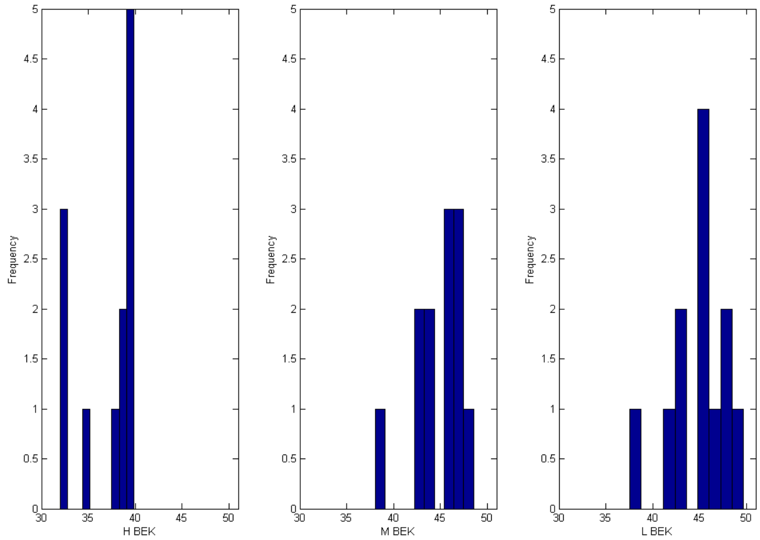

Figure 4.6.: Figure showing the frequency of answers acquired in the Bekesy test. The answers are distributed in ten intervals.

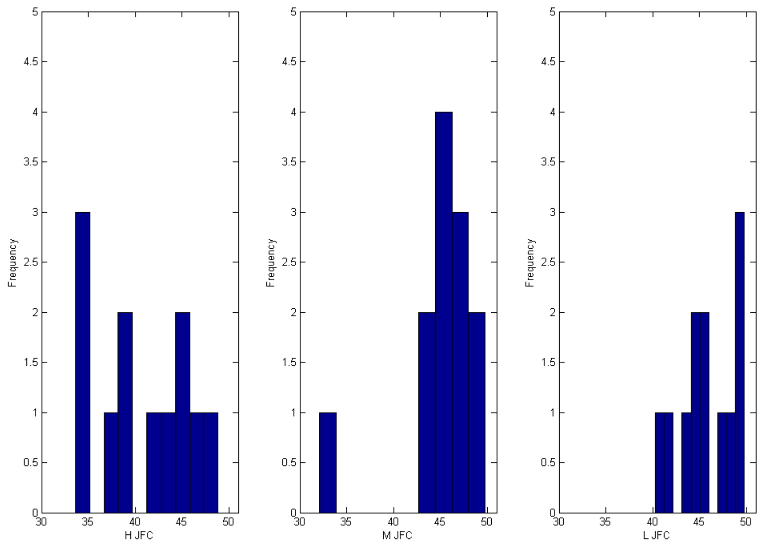

Figure 4.7.: Figure showing the frequency of answers acquired in the JFC test. The answers are distributed in ten intervals.

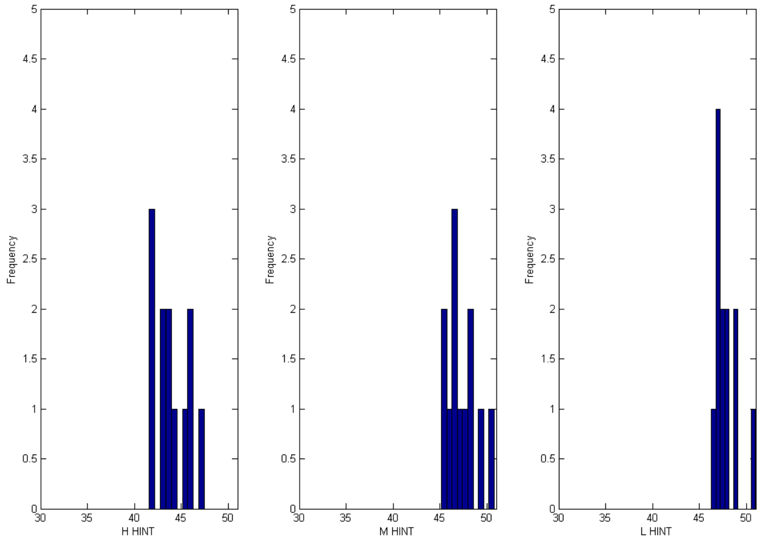

Figure 4.8.: Figure showing the frequency of answers acquired in the HINT test. The answers are distributed in ten intervals.

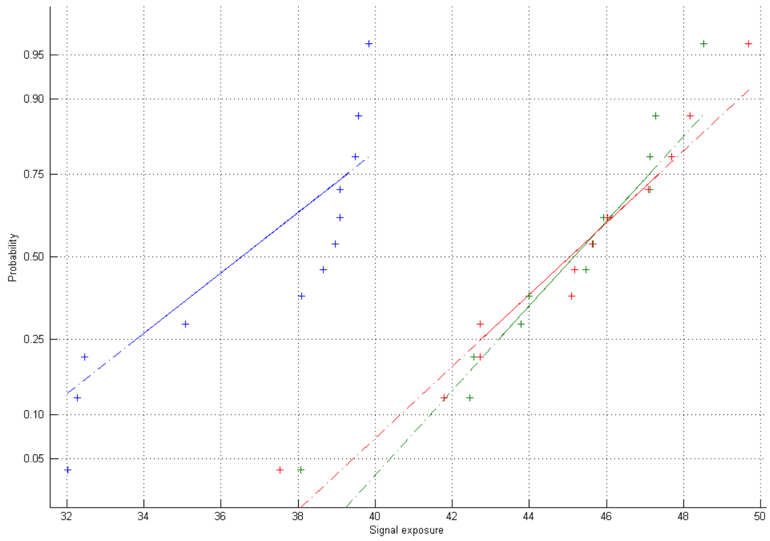

Figure 4.9.: Normality plots of the Bekesy data showing the distribution of data compared to a normal distribution. H noise is blue, M is green and L is red

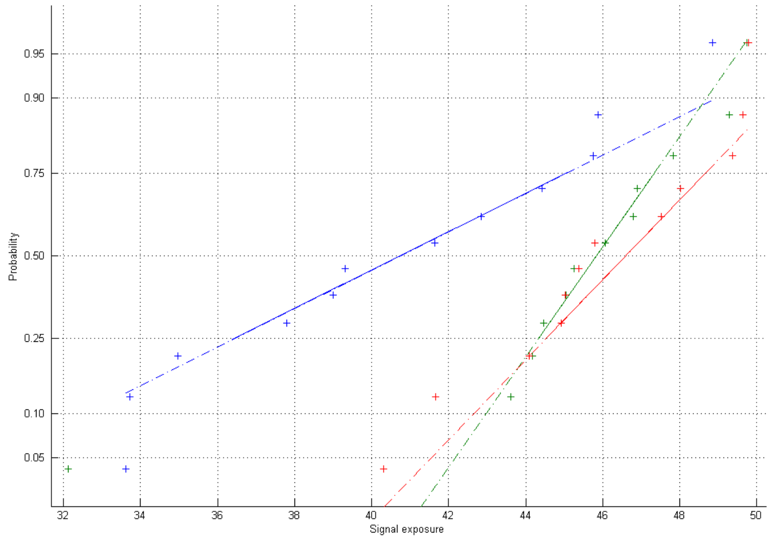

Figure 4.10.: Normality plots of the JFC data showing the distribution. H noise is blue, M is green and L is red

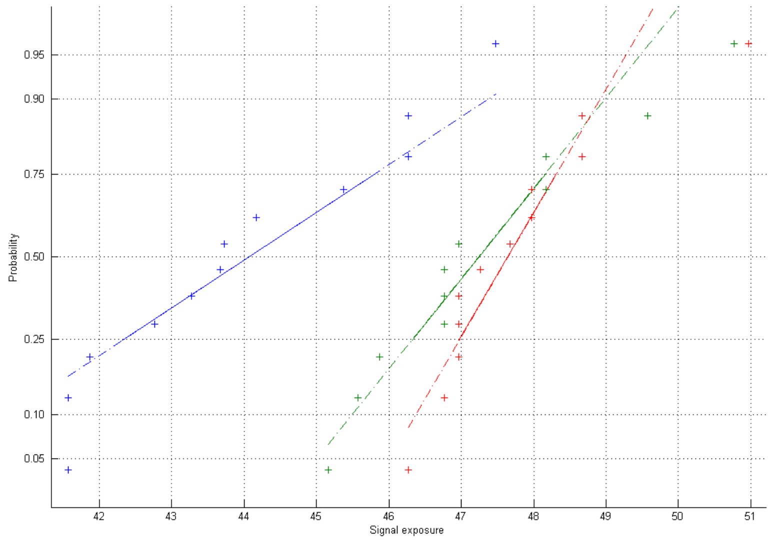

Figure 4.11.: Normality plots of the HINT data showing the distribution. H noise is blue, M is green and L is red

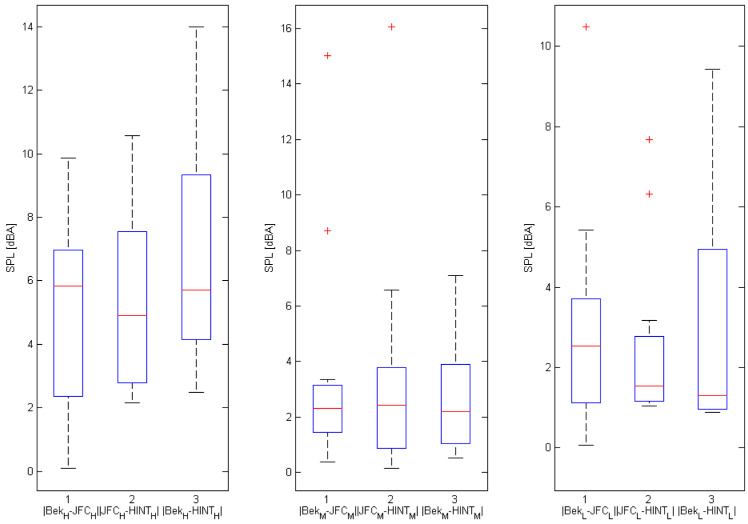

Figure 4.12.: Figure showing the absolute differences between populations tested.

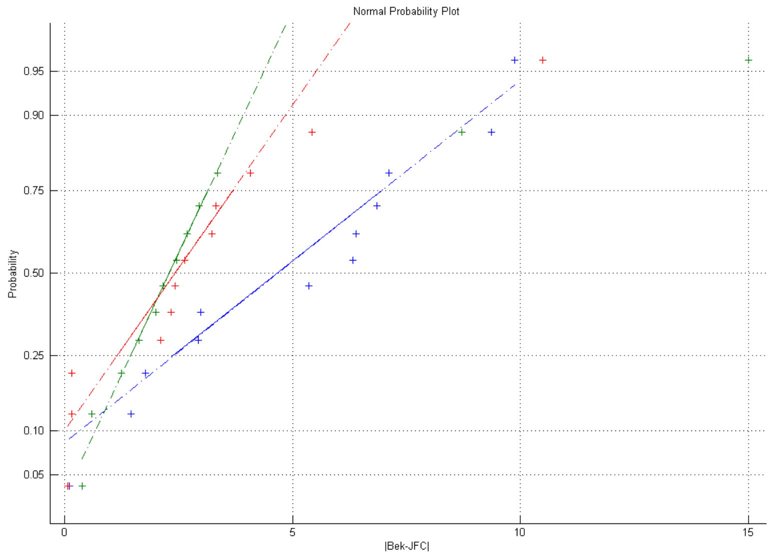

Figure 4.13.: Figure showing the normality distribution of the absolute difference between Bekesy and JFC_A results. Blue is represents H noise, red is M noise and green is L

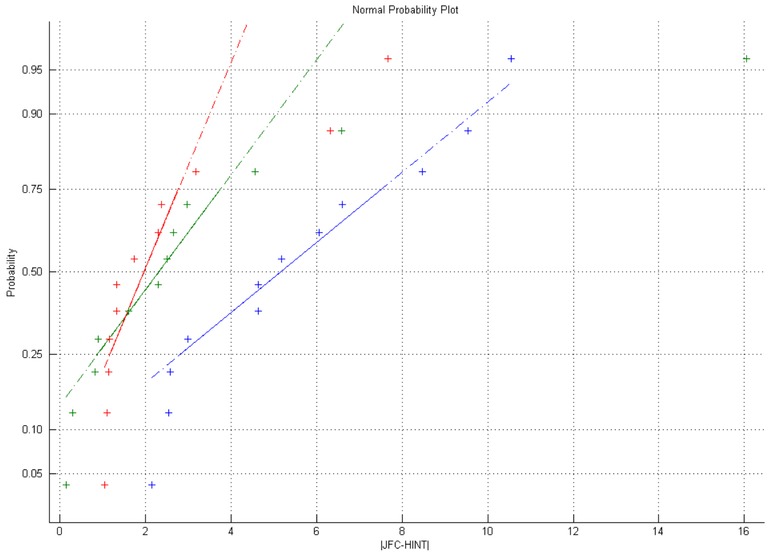

Figure 4.14.: Figure showing the normality distribution of the absolute difference between JFC_A and HINT results. Blue is represents H noise, red is M noise and green is L

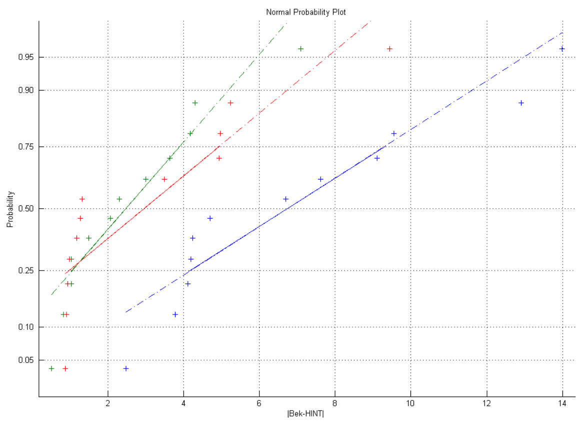

Figure 4.15.: Figure showing the normality distribution of the absolute difference between Bekesy and HINT results. Blue is represents H noise, red is M noise and green is L

4.4.1. Statistical results

The results of the Lilliefors test for normality (conducted in Matlab), are found in table 4.4.

Table 4.4.: Results of Lilliefors test for normal distribution of the absolute differences between data sets.

Data set	H
$BEK_H - JFC_{A,H}$	0
$BEK_M - JFC_{A,M}$	1
$BEK_L - JFC_{A,L}$	0
$BEK_H - HINT_H$	0
$BEK_M - HINT_M$	0
$BEK_L - HINT_L$	1
$JFC_{A,H} - HINT_H$	0
$JFC_{A,M} - HINT_M$	1
$JFC_{A,L} - HINT_L$	1

A T-test for equal means and unknown variance was performed using MatLab. It should be noted that this test assumes that the data comes from a normal distribution, the results presented will be for those distributions who show indications of being normally distributed according to the Lilliefors test. The signal level tests are presented in tables 4.5.

Table 4.5.: Results of T-test for equal means and unknown variance was performed. Distributions who showed normality according to Lilliefors test.

Test variables	H	P	$CI_{95\%}$
$S_{BEK,H}, S_{JFC,H}$	1	0.0263	$\{-6.6836 \leq \mu \leq -0.50823\}$
$S_{BEK,H}, S_{HINT,H}$	1	5.0×10^{-5}	$\{-9.3343 \leq \mu \leq -4.5625\}$
$S_{JFC,H}, S_{HINT,H}$	0	0.0532	$\{-6.7608 \leq \mu \leq 0.055759\}$
$S_{BEK,M}, S_{HINT,M}$	1	0.0012	$\{-3.8429 \leq \mu \leq -1.2446\}$
$S_{BEK,L}, S_{JFC,L}$	0	0.4156	$\{-3.6387 \leq \mu \leq 1.6181\}$

5. Discussion

The speech material chosen, was done so at random. The long time averages between the speech files and the noise files are supposed to be the same but there are slight differences between the files. This was however expected and accepted as it was clear that there would occur a loss of control over the contents of the speech files used. The biggest discrepancies are found between file T08 and noise L (-17.8vs. -14) and T13 and M(-14.4 vs. -17.8).

The STI measurements established that the loudspeakers are in fact capable of transmitting a signal in a normal environment without affecting the intelligibility of it. The STI rating of 0.967 (0.936modified) and the A_{Lcons} of 0.9% yields a rating of Excellent. The frequency response of the loudspeaker is presented in the same region as the STI measurement is made. Judging from figure 4.2 there will be a drop in the response after 2000 Hz. An important frequency band for speech frequencies is found between 500 and 4k Hz and so this drop will have some effect on the reproduction of speech in the region.

As can be seen in chapter 4, a first glance at the results indicates that there is indeed a difference in how well the different methods predict the signal exposure needed for intelligibility. A closer look on table 4.3 shows that the Bekesy test in general predicts the lowest mean exposure for all noise types. The HINT predicts the highest in all cases. Between noise types within a method, there is always at least one dBA difference between the noise types, except for the case of JFC_A and HINT with M and L noise. The JFC_A method does not seem to make independent predictions between M and L noise types, with values of 45.11 dBA and 45.96 dBA and neither does the HINT which gives, for M 47.38 dBA, and for L it gives 47.76 dBA.

In figure 4.3 one can see how the development of the signal exposure level increases with the noise. Since the masking effect increases with lower frequency content, it should by right increase the signal exposure level needed for intelligibility.

The box plots seen in figures 4.4 and 4.5 show how the results are distributed along with the extreme values, which are not considered to be outliers, of each data set. Note how the median values in figure 4.4 suggest that there is not much difference between M and L noise for the any of the methods. Figure 4.5 suggests that all three methods did not differentiate between results gathered for the L noise and give similar results. Preparing for the T-test, an investigation of the normality of the data was conducted.

An investigation of the normal distribution of the absolute difference between data sets was done by normality plots and complemented with the Lilliefors test. The normality plots seen in figures 4.13, 4.14 and 4.15 showed how closely the data follows a normal distribution. Some obvious outliers were found in the far ends of the distributions. Should the data follow a normal distribution, most of the data points would fall on, or very close to the solid or dashed lines seen in the figures.

The hypothesis of normality was tested using the Lilliefors test (results are found in table 4.4), which shows that five out of nine can be seen as normally distributed, and these five cases are the ones which the predictions can therefore be based on. The hypothesis is rejected for the remaining four cases.

A T-test was performed to see whether the the null-Hypothesis would be rejected or if it would fail to be rejected. That is to say: Do the methods differ enough from each other so that one can say that the results are independent? The hypothesis tested was whether two data sets could be regarded as being from normal distributions with equal means, with unknown but equal variances. Results seen in table 4.5 indicate which populations can be used to make predictions. As for the others, a larger population is needed so that predictions can be made on normality distributions.

It seems that all three methods can be used in combination to predict exposure levels for H type noise and that one may be able to use Bekesy and HINT for M-noise predictions independent of each other. Others fail to be of use when predicting exposure levels needed for intelligibility, due to either failure to reject the hypothesis(See table 4.5) or due to lack of normally distributed data which renders the T-test useless.

In figure 4.9 one can see what noise types are distinct with regard to prediction of signal exposures. This is established by the T-test performed, the Bekesy test with H noise is independent of the all other noise types using any method. Note however that the normality was rejected for that case which could have influenced the results of the T-test.

The JFC_A method, seen in figure 4.10, shows however that it is unable to distinguish between the data sets of M and L noise as expected. This indicates that the JFC_A method is not good enough to give reliable predictions of signal exposure with regard to these two noise types. This is further strengthened by the results seen in table 4.3 where the mean exposure predicted is within one dBA between M and L noise.

For the HINT method, the results in figure 4.11 are similar to the JFC_A method. There is no discrimination between results gathered from M and L noise. The results are better for the H noise which is deemed independent of both M and L noise.

6. Conclusion

The goal of the thesis was to develop a system that tested speech intelligibility of hearing protection products with embedded loudspeakers. The main objectives to be completed were the following: The test should be independent from learning, automated to some extent, have the possibility to test different noise types, to be fairly short in time, and to be LabView based.

To achieve the independence of learning, commonly available material was used- namely audiobook tracks. The abundance of audiobooks available takes care of this aspect perfectly. The main concern in choosing the material is to make sure that the recording is done by a professional, and that the quality is acceptable with regard to reverberation and theatrical reading. When developing the LabView program responsible for data collection, it was ensured that a single person (i.e. the test subject) could in fact start the test, take position and carry out the test as a listener. This places some responsibility on the test subject, not to select speech material that one is familiar with. Loading and playing different noise types is controlled by the program as well. For a single noise type and a single test, the total time that it takes to complete one measurement is around six minutes. When collecting data, two tests were run, each with three different noises and so the total time a subject took to finish was roughly 40 minutes.

The results gathered from the tests were analyzed statistically using a pairwise T-test for an equal mean on absolute differences between populations that showed normal distribution according to the Lilliefors test. The results from the T-test showed that in four cases, the data sets gave results that enabled the H_0 hypothesis of equal means to be rejected. In one case the test failed to reject the hypothesis, suggesting a dependency on some underlying variable.

The within-method test between noise types found that in most cases the failure to reject the H_0 hypothesis happened when comparing M and L noise types. No single test gave good results when compared within-method with regard to all noise types. The test managed however to reject between-method data sets between Bekesy and HINT of the noise types H and M. This indicates that for a final test setup, a combination of all three tests should be used to acquire independent data sets to judge the loudspeaker by.

One could use these results in quantifying the signal exposure that gives an intelligible signal and I would propose, judging by the results, that one used Bekesy in combination with JFC_A and HINT method along with H noise, and the Bekesy in combination with the HINT method for M noise. This conclusion is based on table 4.5 in chapter 4.

In general one can say that the T-test results suffer from the small data set. The subjects were only 12 with complete data across all the methods, and even though the T-test is often referred to as a small sample statistical test, it requires at least 20 subjects for reasonably accurate results- the more the better as it influences the width of the CI and the probability of falsely failing to reject a hypothesis or, failing to reject it when it should by right be rejected.[22] It can not be emphasized enough that a sufficiently large population should give a much better result and should also give a more exact distribution with regard to normality, possibly enabling more methods to be combined for prediction of the loudspeaker quality.

7. Future Work

The results put forth in this report show each of these methods succeed in predicting a signal exposure level but statistical results do not indicate that they can be used as a definite measure of intelligibility in an arbitrary test scenario. Since the population tested was small, the next natural step would be to gather more results and process the data again to see whether the results improve.

Setting up a new test scenario one might find it interesting to see if the results are repeatable with a limited population when testing in the same manner as before, possibly with new speech material. The next logical test would be to see if the methods can be used to make successful predictions on an equivalent product and then to see if the test can be applied using an arbitrary noise that is recorded in situ. As mentioned in the introduction chapter, testing might be done on products that rely upon noise canceling, possibly extending the use of the testing method into the area of electronics verification.

The effect that language has on the intelligibility would also be an interesting test to perform. It is however imperative that this be done in a country where the language to be tested is spoken. For example, conducting an English language test on a native Swedish speaker would not be a good idea- Even though the introduction of English language is made early in schools it is no guarantee that people attain the same proficiency(which they almost never do) as a native speaker has, and the lack of proficiency will introduce an unnecessary variable to take into account.

Regarding the development of the LabView programs and the Matlab scripts, there are many possible ways to improve both parts but I will not try to suggest in what way. I have done the Matlab programming to the best of my abilities given the time I had, and for the LabView part I received much help and support.

The philosophy of intelligibility tests is a great one. There are a myriad of tests, variations of tests, objective and subjective, that can be applied in all types of situations to try and give a quantitative or qualitative interpretation of intelligibility. The main problem, as it appears from my point of view, is that people are not sure of what it is that actually makes a signal intelligible. The complex psychological and physical process of understanding a given signal may well be one that is impossible to quantify as of yet. Underlying variables that have not seen the light of day can be a heavy influence on results gathered in all available tests. Further research in this area is therefore needed before any solid and infallible results can be found for intelligibility.

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A. Matlab Scripts

A.1. Function of scripts and script hierarchy

After that data has been acquired from the LabView program, the data is run through a Matlab script. The scripts have a main goal to calculate an average signal exposure level from the LabView data.

A.1.1. RUN_ANALYSIS.m

```
%Run it all
edit ResultFile.dat
edit Thirdoctavedata.txt
pause
Main;

[S_out]= Expo_s(S_full_dB, Locs_maxima,dt,'Thirdoctavedata.txt',[1:length(Locs_maxima)]);

[S_out]= Expo_s(S_full_dB, Locs_minima,dt,'Thirdoctavedata.txt',[1:length(Locs_minima)]);

SNR_srv;

%[EOS]
```

A.1.2. Main.m

```
% function Main
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%   Script loads the results from a single result file
%
%   Usage: Place the script in the folder where your data is collected. The
%   script loads the ResultFile.dat  using the filename as variable name.
%   The script then proceeds to process the data.
%
%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
clear all

% Change name to fit your resultfile name
load ResultFile.dat
% Equivalent level file, make sure this one goes with the right ResultFile!
fid = fopen('Thirdoctavedata.txt');
C = textscan(fid,'%d%f32');
fclose(fid);

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Read the data from the result file to data vectors, perform basic
%calculations.
T_full= ResultFile(:,1);
dt= T_full(2)-T_full(1);                                     %Timestep [s]
S_full= ResultFile(:,3);
% N_full=ResultFile(:,4);
N_full= 80;
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

%Choose the method you used to collect the results.
k1=menu('Method used?','JFC','BEK');

% Sound device data, collected by the measurement system in Lab @the company
L=[20.4000   30.6000   39.7900   50.0000  55.1 57 60.2000 62.25   70.4000
80.6000   89.8000  100.0000];    %[Level]
V=[0.0017    0.0099    0.0334  0.15 0.167  0.0943   0.2110  0.248  0.4210
0.7490    1.0500    1.4110];               %[V]
Comp=4; % Compensating the file used for sound dev. measurements [dB]
```

```
%Create a vector and round it off to two decimals
xi= linspace(0,100,10000);
xi=round(xi*100)/100;
V_interp=interp1(L,V,linspace(0,100,10000),'cubic');

%Mapping of signal level to voltage out
k=0;
for ii=1:length(S_full)
    k=k+1;
    Index_S(k)=find(round(S_full(ii)*100)/100==xi');
    %S_Mapping(k)= V_interp(Index_S)
end

k=0;
for ii=1:1:length(Index_S)
k=k+1;
    Voltage_mapping(k)= V_interp(Index_S(ii));
end

%--------------Voltage mapping and dBV calc---------------------
v_ref= 1; % Max out(Volt_rms) from sound device [V]
Signal_Voltage=Voltage_mapping;
S_full_dB= 20*log10(Signal_Voltage/v_ref)+Comp;    % Level [dB rel. v_ref].

Voltage_dB= 20*log10(V_interp/v_ref)+Comp; % [dB rel v_{ref}] Gives a 0-100 scale

% Compensation is for file used while measuring voltage out.
%--------------------------------------------------------------------------------

% Creates a pseudo-SNR vector for in-script purposes
rSNR= S_full./N_full;

%locations of the peaks you wish to analyze
[X]=ChoosePeaks(T_full,S_full);

        if k1==1 %JFC
                [Peaks_maxima Locs_maxima]=BreakPoint(S_full,rSNR,X);
              Locs_minima=1:1:length(Locs_maxima);
                Locs=[Locs_minima' Locs_maxima];
```

```
        elseif k1==2 %BEK
               %Finds all the peaks in the interval.
               [Peaks_maxima,Locs_maxima]= findpeaks(S_full);
               k=0;
               Locs_minima=length(length(Locs_maxima)-1);
               for ii=1:1:length(Locs_maxima)-1
                   k=k+1;
                   Locs_minima(k)=find(S_full(Locs_maxima(ii):Locs_maxima(ii+1))
               end

               Locs_minima= (Locs_minima-1)+Locs_maxima(1:end-1);
             Locs=[Locs_minima 0; Locs_maxima];
       end

close all
% [EOS]
```

A.1.3. BrakePoint.m

```
function [P BP]=BreakPoint(S_full,rSNR,X)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% This script was written as the function findpeaks() could not
% handle the plateau that  appears in this vector.
%
%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
dSNR= diff(rSNR);
% [M,N]=size(rSNR);
BP= zeros(length(X)/2,1);
P= zeros(length(X)/2,1);
k=0;

    for ii=1:2:length(X)-1
        k=k+1;
   %Finds the last instance of a non-zero value in the interval in X
        BP(k,1)=X(ii)+find(dSNR(X(ii):X(ii+1)),1,'last');
        P(k,1)=S_full(BP(k,1));
    end


end
%[EOF]
```

A.1.4. averageFunction.m

```
function [med_Signal med_SNR]=averageFunction(S,SNR,Locs, k)
%
% average_SNR= zeros(size(max_SNR));

BP_L=Locs(:,1);
BP_H=Locs(:,2);

NN=5;

%Note: If JFC only let it be BP(ii)-NN, adjust the denominator accordingly
    if k==1
```

```
        for ii= 1:length(BP_H)
            %JFC
            med_SNR= median(SNR(BP_H(ii)-NN:BP_H(ii)));
             med_Signal= median(S(BP_H(ii)-NN:BP_H(ii)));
        end
    elseif k==2
        %Bek
        Median_low= median(SNR(min(BP_L):max(BP_L)));
        Median_high= median(SNR(min(BP_H):max(BP_H)));
       med_SNR= (Median_low+Median_high)/2;

       Median_low= median(S(min(BP_L):max(BP_L)));
        Median_high= median(S(min(BP_H):max(BP_H)));
       med_Signal= (Median_low+Median_high)/2;
    else
      disp('unknown');
    end

end
%[EOF]
```

A.1.5. Expo_s.m

```
function [S_out]= Expo_s(S, Locs_s,dt,oct,file)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% Calculate the level from octaveband analysis from the chosen
% soundsnippets, requires LabView file "Thirdoctavedata.txt".
%
% = brakepoints (peaks)
% S= S_full_dB, Locs_s= Locs_maxima || Locs_minima dt= Timestep
%
%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

men=menu('Max  /  Min?', 'Max', 'Min');
fid = fopen(oct);
C = textscan(fid,'%s%f32');
fclose(fid);
TROD=C;
```

```
Index_ofCP=zeros(length(file),1);
Value_ofTROD=zeros(length(file),1);
S_dB_SPL=zeros(length(file),1);
k=0;
    for ii=1:length(file)
            k=k+1;
                sint=file(ii);
                %------------------------------
             %Note that the vectors are indexed backwards because of the way the sound cl
             % numbered.
             Locs_sorted=sort(Locs_s,'descend');
             Index_ofCP(k)= Locs_sorted(sint); %Peak in clip, location of
            % Index_ofTROD= find(TROD{1}(:)==sint) % Searches for the index of the value
             % Thirdoctavedata value. Used to calculate the exp. value.
            Value_ofTROD(k)=TROD{2}(sint);
            %------------------------------

            %------------------------------
            %Ensure this value is the same as in the LabView part where signal clip is
            %produced +/- time around the brakepoint. In any case, this should be half
            %the length of the signal clip. Rounded off downwards.
             DT=dt;
            %DT=0.06;
            Steps= floor(1/DT);
            %This gives the signal part where the clip is located.
            % S= S(Index_ofCP-Steps:Index_ofCP+Steps);

            %Gives the dBV value at peak
            S(k)= S(Index_ofCP(k));

            % Clip's exposure values dBA, S=dBV voltage of clip Value_ofTROD= thirdoctave
            S_dB_SPL(k)= S(k)+ Value_ofTROD(k);
    end
S_out= S_dB_SPL;
%------------------------------

    if men==1
        fp=fopen('Clip_high.txt','w');
        fprintf(fp,'%f \n', S_out);
```

```
        fclose(fp);
    elseif men==2
        fp=fopen('Clip_low.txt','w');
        fprintf(fp,'%f \n', S_out);
        fclose(fp);
     end

end
     %[EOF]
```

A.1.6. SNR_srv.m

```
k2=menu('Noise type','H','M','L');
pref=20e-6;
if k2==1
N_dBA= 86.15;%Level H [dBA] with off. compens.
p_n=10^(N_dBA/20)*pref;
Noise='H';
elseif k2==2
N_dBA= 81.65;   %Level M [dBA]with off. compens
p_n=10^(N_dBA/20)*pref;
Noise='M';
elseif k2==3
N_dBA= 80.93;   %Level L [dBA]with off. compens.
p_n=10^(N_dBA/20)*pref;
Noise='L';
end

if k1==1
        fp=fopen('Clip_high.txt','r');
        Signal=fscanf(fp,'%f',inf);
p_rms= 10^(Signal/20)*pref;
elseif k1==2
        fp1=fopen('Clip_high.txt','r');
        Signal_h=fscanf(fp1,'%f',inf);
        fp2=fopen('Clip_low.txt','r');
        Signal_l=fscanf(fp2,'%f',inf);
        Signal= (median(Signal_h)+median(Signal_l))/2;
        p_rms= 10^(Signal/20)*pref;
```

```
end

%Signal[dBA]-Noise[dBA]
 SNR_dB=Signal-N_dBA;
 SNR=10*log10(p_rms/p_n);

 SNR_dBmean= mean(SNR_dB);
 SNR_mean= mean(SNR);

 SNR_dBstdv= std(SNR_dB);
 SNR_stdv=std(SNR);


 %Note that the trimmean function trims off 10% of the extreme values.
Signal_av=mean(Signal); % average [dBA]

fp=fopen('Analysis_Results.txt','w');
fprintf(fp,'%s %f %f %f \n',Noise, N_dBA, Signal_av, SNR_mean);
fclose(fp);

%[EOS]
```