2001

DSP algorithms for digital hearing instruments.

Erkan. Onat

University of Windsor

Follow this and additional works at: https://scholar.uwindsor.ca/etd

Recommended Citation


https://scholar.uwindsor.ca/etd/1786

This online database contains the full-text of PhD dissertations and Masters' theses of University of Windsor students from 1954 forward. These documents are made available for personal study and research purposes only, in accordance with the Canadian Copyright Act and the Creative Commons license—CC BY-NC-ND (Attribution, Non-Commercial, No Derivative Works). Under this license, works must always be attributed to the copyright holder (original author), cannot be used for any commercial purposes, and may not be altered. Any other use would require the permission of the copyright holder. Students may inquire about withdrawing their dissertation and/or thesis from this database. For additional inquiries, please contact the repository administrator via email (scholarship@uwindsor.ca) or by telephone at 519-253-3000 ext. 3208.
INFORMATION TO USERS

This manuscript has been reproduced from the microfilm master. UMI films the text directly from the original or copy submitted. Thus, some thesis and dissertation copies are in typewriter face, while others may be from any type of computer printer.

The quality of this reproduction is dependent upon the quality of the copy submitted. Broken or indistinct print, colored or poor quality illustrations and photographs, print bleedthrough, substandard margins, and improper alignment can adversely affect reproduction.

In the unlikely event that the author did not send UMI a complete manuscript and there are missing pages, these will be noted. Also, if unauthorized copyright material had to be removed, a note will indicate the deletion.

Oversize materials (e.g., maps, drawings, charts) are reproduced by sectioning the original, beginning at the upper left-hand corner and continuing from left to right in equal sections with small overlaps.

ProQuest Information and Learning
300 North Zeeb Road, Ann Arbor, MI 48106-1346 USA
800-521-0600

UMI®
DSP ALGORITHMS FOR DIGITAL
HEARING INSTRUMENTS

by

Erkan Onat

A Thesis
Submitted to the Faculty of Graduate Studies and Research
through Electrical and Computer Engineering
in Partial Fulfillment of the Requirements for
the Degree of Master of Applied Science at the
University of Windsor

Windsor, Ontario, Canada

2001

© 2001 Erkan Onat
The author has granted a non-exclusive licence allowing the National Library of Canada to reproduce, loan, distribute or sell copies of this thesis in microform, paper or electronic formats.

L’auteur a accordé une licence non exclusive permettant à la Bibliothèque nationale du Canada de reproduire, prêter, distribuer ou vendre des copies de cette thèse sous la forme de microfiche/film, de reproduction sur papier ou sur format électronique.

The author retains ownership of the copyright in this thesis. Neither the thesis nor substantial extracts from it may be printed or otherwise reproduced without the author’s permission.

L’auteur conserve la propriété du droit d’auteur qui protège cette thèse. Ni la thèse ni des extraits substantiels de celle-ci ne doivent être imprimés ou autrement reproduits sans son autorisation.

0-612-75771-4
ABSTRACT

A new digital filter bank design and a new compression algorithm that can improve the performance of hearing instruments located completely in the ear canal (CIC) are developed in the thesis. In order to assess state-of-the-art hearing instruments employing advanced signal processing techniques the DynamEQ-II analog hearing instrument developed by the Gennum Corporation was studied extensively. A sophisticated SIMULINK model, involving the use of audio files, was developed to evaluate the performance characteristics of the strategies and algorithms used in the DynamEQ-II. The RangeEar algorithm employed in the DigiFocus hearing instrument from the Oticon Company was also studied using SIMULINK in a similar manner. Two recommended improvements for a new hearing instrument are presented. The first improvement involves the use of an eight-band digital filter bank based on an interpolated finite impulse response (IFIR) prototype filter that has been optimized using delay elements to give a maximally flat overall magnitude response. The resulting group delay is a constant and less than the value where self-hearing and “lip reading” problems occur. The second improvement uses a new compression algorithm based on a model of the human auditory system. The new algorithm replaces the existing constant homomorphic multiplication algorithms with an acoustic signal intensity weighted multiplication. The resulting nonlinear compression ratio expands low level signals and compresses high level signals in such a manner so as to improve noise immunity and increase the intelligibility of the sound. The MIT hearing loss simulator was employed to evaluate the effectiveness of the new proposed filter bank and compression algorithm by analysis of and listening to actual test audio files.
To My Dear Parents
Acknowledgements

I would like to express my deepest thanks and appreciation, for his continuous technical guidance, support and encouragement, to Dr. W. C. Miller. His wisdom was the light in this research.

I also want to express my appreciation to Dr. G. A. Jullien and Dr. R. G. Gaspar for their interest and invaluable comments. This thesis would not be written without the skillful management of Dr. G. A. Jullien in VLSI Research Group.

Finally, I would like to express my thanks to GENNUM Corporation for their crucial support.
TABLE OF CONTENTS

ABSTRACT iii

DEDICATION iv

ACKNOWLEDGEMENT v

LIST OF TABLES ix

LIST OF FIGURES x

Chapter 1 Introduction 1

1.1 Hearing Instrument Technologies 2

1.2 Hearing Instrument Types 3

1.3 Overview 5

Chapter 2 Human Hearing 7

2.1 The Ear Structure 7

2.2 Limits of Human Hearing 15

2.3 Hearing Loss Types 20

2.4 Discussion on Hearing Instrument Design Constraints 27

Chapter 3 DynamEQ-II Hearing Instrument 31

3.1 Signal Representation 31

vi
<table>
<thead>
<tr>
<th>Chapter</th>
<th>Title</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>3.2</td>
<td>DynamEQ-II Specifications</td>
<td>33</td>
</tr>
<tr>
<td>3.3</td>
<td>DynamEQ-II Digital System Simulator</td>
<td>40</td>
</tr>
<tr>
<td>3.4</td>
<td>Discussion</td>
<td>57</td>
</tr>
<tr>
<td>Chapter 4</td>
<td>Digital Filter Bank Design for Hearing Instruments</td>
<td>59</td>
</tr>
<tr>
<td>4.1</td>
<td>Design Constraints</td>
<td>59</td>
</tr>
<tr>
<td>4.2</td>
<td>IFIR Filter Bank Design</td>
<td>62</td>
</tr>
<tr>
<td>4.3</td>
<td>QMF (Quadrature Mirror Filter) Bank Design</td>
<td>82</td>
</tr>
<tr>
<td>4.4</td>
<td>Wavelet Filter Bank</td>
<td>89</td>
</tr>
<tr>
<td>4.5</td>
<td>DFT Modulated Filter Bank</td>
<td>93</td>
</tr>
<tr>
<td>4.6</td>
<td>Discussion</td>
<td>107</td>
</tr>
<tr>
<td>Chapter 5</td>
<td>Compression Algorithms and Performance Evaluation</td>
<td>108</td>
</tr>
<tr>
<td>5.1</td>
<td>RangeEar Compression Algorithm</td>
<td>108</td>
</tr>
<tr>
<td>5.2</td>
<td>Homomorphic Multiplicative AGC</td>
<td>113</td>
</tr>
<tr>
<td>5.3</td>
<td>Hearing Loss Simulator</td>
<td>123</td>
</tr>
<tr>
<td>5.4</td>
<td>Performance Evaluation of Compression Algorithms</td>
<td>126</td>
</tr>
<tr>
<td>5.5</td>
<td>Discussion</td>
<td>140</td>
</tr>
<tr>
<td>5.5</td>
<td>Overall Discussion</td>
<td>141</td>
</tr>
<tr>
<td>Chapter 6</td>
<td>Conclusions</td>
<td>143</td>
</tr>
<tr>
<td>6.1</td>
<td>Future Work</td>
<td>144</td>
</tr>
<tr>
<td>Section</td>
<td>Title</td>
<td>Page</td>
</tr>
<tr>
<td>------------------</td>
<td>----------------------------------------------------------------------</td>
<td>------</td>
</tr>
<tr>
<td>References</td>
<td></td>
<td>146</td>
</tr>
<tr>
<td>Appendix I</td>
<td>Matlab Code for HRTF Data Plotting</td>
<td>151</td>
</tr>
<tr>
<td>Appendix II</td>
<td>Optimization of Delay Characteristics in IFIR</td>
<td>155</td>
</tr>
<tr>
<td>Appendix III</td>
<td>Matlab Code for Eigenfilter Design</td>
<td>158</td>
</tr>
<tr>
<td>Appendix IV</td>
<td>Simulink Code of Homomorphic Multiplicative AGC</td>
<td>160</td>
</tr>
<tr>
<td></td>
<td>Algorithm with 8-band IFIR Filter Bank</td>
<td></td>
</tr>
<tr>
<td>VITA AUCTORIS</td>
<td></td>
<td>255</td>
</tr>
</tbody>
</table>
# LIST OF TABLES

<table>
<thead>
<tr>
<th>Table 2-1</th>
<th>Sound examples for dynamic range of human hearing</th>
<th>16</th>
</tr>
</thead>
<tbody>
<tr>
<td>Table 2-2</td>
<td>Octave bands of human hearing and their characteristics</td>
<td>17</td>
</tr>
<tr>
<td>Table 2-3</td>
<td>Hearing loss degrees and their effects</td>
<td>20</td>
</tr>
<tr>
<td>Table 2-4</td>
<td>Sensorineural hearing loss with age</td>
<td>23</td>
</tr>
<tr>
<td>Table 3-1</td>
<td>Dynamic Range Comparisons</td>
<td>32</td>
</tr>
<tr>
<td>Table 4-1</td>
<td>Cutoff frequencies of IFIR filter bank channels</td>
<td>69</td>
</tr>
<tr>
<td>Table 4-2</td>
<td>Cutoff frequencies of 60 dB IFIR filter bank</td>
<td>79</td>
</tr>
<tr>
<td>Table 4-3</td>
<td>DFT filter bank implementation cost</td>
<td>105</td>
</tr>
<tr>
<td>Table 5-1</td>
<td>Audiogram values for first part of simulations</td>
<td>129</td>
</tr>
<tr>
<td>Table 5-2</td>
<td>Audiogram values for second part of evaluation</td>
<td>135</td>
</tr>
</tbody>
</table>
LIST OF FIGURES

Figure 1-1  Digital hearing instrument structure  3
Figure 1-2  Hearing instrument types  4
Figure 1-3  CIC Hearing instrument placement  5
Figure 2-1  The main parts of ear anatomy  7
Figure 2-2  The average HRTF in 0°-180° azimuth angle range  9
Figure 2-3  Mesh plot of HRTF Measurements  10
Figure 2-4  Cross Section of Cochlea  13
Figure 2-5  Functional Diagram of Human Ear  14
Figure 2-6  Place Principle of Cochlea  14
Figure 2-7  Frequency and intensity ranges of speech and music  18
Figure 2-8  Fletcher&Munson equal loudness contours  19
Figure 2-9  Audiogram example for conductive hearing loss  22
Figure 2-10  Audiogram example for sensorineural hearing loss  23
Figure 2-11  Noise induced threshold shift  24
Figure 2-12  Ototoxicity hearing loss  25
Figure 2-13  Meniere’s disease hearing loss  26
Figure 2-14  Normal and impaired loudness growth  27
Figure 2-15  Auditory canal compensation  28
Figure 2-16  Compressive amplification  29
Figure 3-1  DynamEQ-II system model  33
Figure 3-2  DynamEQ-II functional block diagram  34
Figure 3-3  Demonstration of attack and release times for DynamEQ-II  36
Figure 3-4  Compression ratio effect on I/O curve  
Figure 3-5  Lower threshold effect on I/O curve  
Figure 3-6  MPO effect on I/O curve  
Figure 3-7  Gain effect on I/O curve  
Figure 3-8  DynamEQ-II digital system simulator  
Figure 3-9  Frequency response of the simulator low pass filter  
Figure 3-10  Frequency response of simulator high pass filter  
Figure 3-11  Frequency response of slow average detector  
Figure 3-12  Frequency response of fast average detector  
Figure 3-13  Average detector structure in DynamEQ-II simulator  
Figure 3-14  dB converter in simulator  
Figure 3-15  Gain control block of DynamEQ-II simulator  
Figure 3-16  Simulated compression ratio change effect on I/O  
Figure 3-17  Simulated lower threshold change effect on I/O  
Figure 3-18  Simulated gain change effect on I/O  
Figure 3-19  Simulated MPO change effect on I/O  
Figure 3-20  Magnitude of the input test signal  
Figure 3-21  Intensity level of input test signal  
Figure 3-22  Simulator response to test signal  
Figure 3-23  Modified simulator response to test signal  
Figure 3-24  Input speech waveform  
Figure 3-25  Histogram of the intensity levels of input speech  
Figure 3-26  High channel I/O curve for speech simulation  
Figure 3-27  Simulator output waveform  

xi
Figure 3-28  Simulator output histogram  56
Figure 3-29  Spectrum of input sound signal  57
Figure 3-30  Spectrum of simulator output  57
Figure 4-1   Block diagram of IFIR technique  62
Figure 4-2   Magnitude response of lowpass filter, which is designed in one step 63
Figure 4-3   Model filter magnitude response  64
Figure 4-4   Zero padded model filter magnitude response  65
Figure 4-5   Magnitude response after image suppression  65
Figure 4-6   Complement of zero padded model filter  66
Figure 4-7   IFIR Filter Bank Structure  68
Figure 4-8   Magnitude responses of pass bands  70,71
Figure 4-9   Phase response of first passband  72
Figure 4-10  Total magnitude response of the filter bank  73
Figure 4-11  The delay characteristics of filter bank channels  73
Figure 4-12  Optimization Process of delay elements  75
Figure 4-13  Maximally flat overall magnitude response  74
Figure 4-14  The effect of 1st channel gain increase on overall response  76
Figure 4-15  The effect of 5th channel gain increase on overall response  77
Figure 4-16  The effect of 7th channel gain increase on overall response  77
Figure 4-17  Magnitude responses of 60 dB IFIR design pass bands  78
Figure 4-18  Overall magnitude response of 60 dB IFIR design  80
Figure 4-19  Maximally flat overall magnitude response for 60dB IFIR design  80
Figure 4-20  The effect of gain increase on 60 dB IFIR filter bank  81
Figure 4-21  Block diagram of QMF bank  82
<table>
<thead>
<tr>
<th>Figure</th>
<th>Description</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>4-22</td>
<td>Impulse responses of the first band of QMF bank</td>
<td>84</td>
</tr>
<tr>
<td>4-23</td>
<td>Gain increase effect on QMF banks overall response</td>
<td>87</td>
</tr>
<tr>
<td>4-24</td>
<td>The block diagram of extended QMF bank</td>
<td>88</td>
</tr>
<tr>
<td>4-25</td>
<td>Passbands of extended QMF bank</td>
<td>88</td>
</tr>
<tr>
<td>4-26</td>
<td>Dyadic analysis filter bank</td>
<td>90</td>
</tr>
<tr>
<td>4-27</td>
<td>Dyadic synthesis filter bank structure</td>
<td>91</td>
</tr>
<tr>
<td>4-28</td>
<td>Dyadic filter bank structure in simulink for wavelet gain test</td>
<td>91</td>
</tr>
<tr>
<td>4-29</td>
<td>Dyadic discrete meyer wavelet filter bank magnitude responses</td>
<td>92</td>
</tr>
<tr>
<td>4-30</td>
<td>Overall Magnitude response of discrete meyer wavelet filter bank</td>
<td>93</td>
</tr>
<tr>
<td>4-31</td>
<td>Multirate filter bank block diagram</td>
<td>94</td>
</tr>
<tr>
<td>4-32</td>
<td>Complex modulation of a FIR filter</td>
<td>95</td>
</tr>
<tr>
<td>4-33</td>
<td>Modulated filter bank structure</td>
<td>96</td>
</tr>
<tr>
<td>4-34</td>
<td>Noble identities</td>
<td>97</td>
</tr>
<tr>
<td>4-35</td>
<td>DFT filter bank block diagram</td>
<td>98</td>
</tr>
<tr>
<td>4-36</td>
<td>DFT Filter bank simulator in Simulink</td>
<td>101</td>
</tr>
<tr>
<td>4-37</td>
<td>Magnitude response of the analysis filter</td>
<td>102</td>
</tr>
<tr>
<td>4-38</td>
<td>Magnitude response of synthesis filter</td>
<td>103</td>
</tr>
<tr>
<td>4-39</td>
<td>The effect of gain increase on DFT filter bank overall magnitude response</td>
<td>104</td>
</tr>
<tr>
<td>4-40</td>
<td>Overall magnitude response of the DFT filter bank</td>
<td>105</td>
</tr>
<tr>
<td>4-41</td>
<td>Group delay of the DFT filter bank with no gain</td>
<td>106</td>
</tr>
<tr>
<td>4-42</td>
<td>Group delay of the DFT filter bank at 30 dB gain in first channel</td>
<td>106</td>
</tr>
<tr>
<td>5-1</td>
<td>RangeEar system block diagram</td>
<td>109</td>
</tr>
<tr>
<td>5-2</td>
<td>Magnitude responses of RangeEar filter bank</td>
<td>109</td>
</tr>
</tbody>
</table>
Figure 5-3  I/O curve of loudness growth function
Figure 5-4  RangeEar System Simulator in Simulink
Figure 5-5  RangeEar low-frequency compressor block
Figure 5-6  RangeEar compression algorithm I/O response with increasing HTL
Figure 5-7  Loudness perception model
Figure 5-8  The block diagram of homomorphic multiplicative AGC for one pass band
Figure 5-9  Homomorphic multiplicative AGC Hearing Instrument Simulator
Figure 5-10 Homomorphic multiplicative AGC compressor block
Figure 5-11 The effect of HL increase on homomorphic I/O curve
Figure 5-12 The effect of UCL increase on homomorphic I/O curve
Figure 5-13 The effect of NHT increase on homomorphic I/O curve
Figure 5-14 Improved homomorphic multiplicative AGC I/O curve
Figure 5-15 Comparison of the homomorphic algorithms in terms of input, output voltages
Figure 5-16 Compression ratios of I/O curves of Fig. 5-14
Figure 5-17 Improved homomorphic multiplicative AGC algorithm
Figure 5-18 $T_x$ parameter effect on loss simulator I/O curve
Figure 5-19 $T_i$ parameter effect on loss simulator I/O curve
Figure 5-20 Magnitude responses of loss simulator filter bank channels
Figure 5-21 Hearing Loss Simulator in Matlab-Simulink
Figure 5-22 Performance evaluation set-up
Figure 5-23 First hearing loss audiogram for performance evaluation
Figure 5-24  Time waveform and spectrum of the speech signal
for performance evaluation 129

Figure 5-25  Speech waveform envelope extracted as average amplitude 130
Figure 5-26  Speech spectrum envelope extracted as average magnitude 130
Figure 5-27  RangeEar time waveform envelope for speech signal 131
Figure 5-28  RangeEar spectrum envelope for speech signal 132
Figure 5-29  Homomorphic multiplicative AGC time waveform envelope
for speech signal 133
Figure 5-30  Homomorphic multiplicative AGC spectrum envelope
for speech signal 134
Figure 5-31  Second hearing loss audiogram for performance evaluation 135
Figure 5-32  Time waveform and spectrum of the music signal
for performance evaluation 135
Figure 5-33  Music time waveform envelope 136
Figure 5-34  Music spectrum envelope 136
Figure 5-35  RangeEar time waveform envelope for music 137
Figure 5-36  RangeEar spectrum envelope for music signal 138
Figure 5-37  Homomorphic multiplicative AGC time waveform envelope
for music signal 139
Figure 5-38  Homomorphic multiplicative AGC spectrum envelope
for music signal 139
Chapter 1

INTRODUCTION

In this thesis research on DSP algorithms for CIC (completely in the canal) digital hearing instrument is carried out. The DSP algorithms covered are digital filterbanks and digital compression algorithms. Digital version of state of the art DynamEQ-II hybrid hearing aid is simulated. Current technology digital filterbank designs and compression algorithms are simulated and improved. Performances are compared with the aid of a hearing loss simulator.

The history of hearing instruments is tracked from 19th century on, but it had been known that Greeks used shells and Romans had bronze funnels as hearing instruments. In 1800 the first hearing instrument company, which was manufacturing tubes and trumpets as hearing instruments was established [1]. In mid 1890's the first patent for electrical hearing aid is filed but never reached to production. 1899 was the year for the first commercially manufactured hearing aid called Akoullalion [2], which was made of carbon. The first vacuum tube hearing aid, which was very heavy and worn on the body, was patented in 1921. It was consisting of microphone, earphone, amplifier and two batteries, which would last only a day. With the introduction of transistor technology after 1953 the size of the hearing aids got smaller and their capabilities were increased. In 1970 the first hybrid hearing aids, which had analog and digital circuitry were manufactured [2]. In 1988 many programmable hearing aids were introduced. They had
analog circuitry, which was programmed with computers digitally. In 1996 the first behind the ear 100% digital hearing instrument was manufactured.

1.1 Hearing Instrument Technologies

As listed in the history of hearing instruments there has been a transition from analog technology to digital technology. This transition brought three major types of hearing instruments:

**Analog Hearing Instruments:** This type of hearing instruments is also called the conventional hearing instruments. They consist of the microphone, the amplifier unit, the loudspeaker and the battery. The circuitry used in these hearing instruments is totally analog. There are several drawbacks of using the analog technology: The high resolution in frequency domain achieved with digital technology can not be achieved. Analog hearing aids have up to two frequency channels [3]. They are amplifiers, whose electroacoustical performances are adjusted manually through trimmer potentiometers. They apply linear amplification.

**Programmable Hearing Instruments:** This type of hearing instruments is based on analog circuitry, where a memory module replaces the potentiometers. The memory module can be a RAM or EEPROM, which are accessed through an external microprocessor. They are basically analog hearing aids, whose control is improved through digital technology. Even if the patients hearing changes they can be reprogrammed instead of having to obtain a completely new instrument. Compression can be achieved through the sound-level dependent amplification [4].
Digital Hearing Instruments: They convert the analog signal received from the microphone to digital and process the signal totally in digital domain. As demonstrated in Fig. 1-1 the digital signal is processed with DSP algorithm and the output is converted again to analog signal, which drives the loudspeaker. In signal processing part the sound is splitted into multichannels, where the frequency shaping of the sound signal is done according to the patients audiogram. Nonlinear amplification parameters are set once and the algorithm adjusts itself according to the intensity level of the input sound. Besides compression algorithms speech enhancement algorithms are also used [5]. It allows better overall sound quality and compensation of loudness growth.

Fig. 1-1 Digital hearing instrument structure

1.2 Hearing Instrument Types

Hearing instruments can be grouped according to their sizes into four categories: BTE (behind the ear), ITE (in the ear), ITC (in the canal) and CIC (completely in the canal). Their location on the ear is plotted in Fig. 1-2 [6].

BTE (Behind the ear): These hearing instruments rest behind the earlobe and connected to the ear by a custom earmold. Their location allows relaxation in their size constraint.
Therefore circuits allowing more gain can be used. It can be used for a wide range of hearing losses from mild to profound.

![Hearing instrument types](image)

**Fig. 1-2 Hearing instrument types**

**ITE (In the ear):** They fit in the bowl of the ear and are visible in the ear. They are also powerful based on their size and can be used for hearing losses from mild to severe.

**ITC (In the canal):** This hearing instruments are located more in the canal, but still visible. They can be used for hearing losses from mild to moderate.

**CIC (completely in the canal):** They go very deep inside the ear canal and are almost invisible as shown in Fig. 1-3 [7]. Therefore they have more comfort and cosmetic value. Because of the closer proximity to the eardrum and the resonance characteristic of the ear canal, less power is required to provide equal amount of amplification. Other hearing aids have the 'occlusion effect', which means the sensation of talking in a barrel. This is
greatly reduced or eliminated with CIC hearing aid [8]. Another advantage is the reduced wind noise.

![CIC Hearing instrument placement](image)

**Fig. 1-3 CIC Hearing instrument placement**

### 1.3 Overview

The first chapter gives an introduction into thesis by explaining the hearing instrument history, technology and types. It points out that the research covers filterbank designs and compression algorithms for CIC digital hearing instruments, which are tested with hearing loss simulator.
Second chapter explains the anatomy of human hearing and relates the limitations of human hearing and characteristics of hearing losses with hearing instrument design constraints.

Gennum Corporation's DynamEQ-II hybrid hearing instrument characteristics are explained in third chapter. The research on its digital version simulation is explained and the limitations are pointed out.

Chapter four covers the research on digital filterbank design for hearing instruments. Research on interpolated FIR filterbank design, modulated filterbank design, extension of QMF (Quadrature mirror filter) to multichannels and wavelet filterbank design are explained. Their suitability for hearing instruments is discussed.

Two state of the art compression algorithms are demonstrated in chapter five. Their Simulink implementations are explained and performances are evaluated using the hearing loss simulator. The preference is discussed.

Summary of the research and future research possibilities are given as the conclusion in chapter six.
Chapter 2

HUMAN HEARING

In this chapter the mechanism of human hearing is explained. Important characteristics of human hearing for hearing instrument design are clarified. Different types of hearing losses and their effects are pointed out. Hearing instrument design constraints are discussed.

2.1 The Ear Structure

As one of the most intricate and delicate mechanical structures in the human body, the human ear can be divided into three main parts: Outer, middle, and inner ear Fig. 2-1 [9]. In this section the working principles and characteristics of these parts are explained.

Fig. 2-1 The main parts of ear anatomy
2.1.1 Outer and Middle Ear

The outer ear is composed of two parts, the pinna and the ear canal. The pinna of the outer ear serves as a horn collecting sound into the canal. In terms of amplification human pinna is relatively ineffective comparing to some animals pinna, which supplies an appreciable gain over certain frequency ranges [10]. However, this part of the ear helps in localizing the direction of the sound source. The time delay to each ear from sounds on the right or the left as well as sounds from the front and the back is enhanced by the pinna [11]. The ear canal is an approximately straight tube, about 0.7 cm in diameter and 2.5 cm long. At the resonance of the canal, which is about 3 kHz, the SPL at the drum is about 10 dB higher than at the entrance of the canal [10]. The resonance curve of the canal is quite broad. Therefore some gain is observed over the frequency range from 2 to 6 kHz.

Measurements have been done to obtain the average pressure transformation from the free field to the eardrum [14]. The data pool covers 100 subjects in five countries over a period of 40 years. The data is showing the average pressure transformation from the free field to the human eardrum in the horizontal plane as a function of frequency at 15° intervals in azimuth. That means the sound source is aligned with the eardrum in vertical axis and the measurement is performed at every 15° angle increment on horizontal axis with constant distance from the drum. 0° angle corresponds to facing the source. At \( f \) frequency and \( \theta \) degree the measurements are done both in the eardrum \( L_a(f, \theta) \) and at the same point when the subject is removed \( L_r(f, \theta) \). The sound pressure transformation \( T_a(f, \theta) \) is given as:
\[ T_d(f, \theta) = L_d(f, \theta) - L_f(f, \theta) \] (2-1)

These measurements give us the average frequency response of the outer ear with combination of the diffraction of sound waves by the head and shoulders. This total response is called as HRTF (Head related transfer function). Fig. 2-2 is showing the average HRTF in the range of 0° to 180°. Fig. 2-3 is the mesh plot of the total data, showing the change of the HRTF with respect to measurement frequency and angle. From Fig. 2-2 it is observed that the frequency ranges 2 kHz-6 kHz and 11 kHz-12 kHz are more emphasized by the HRTF. The maximum difference of emphasis at different frequencies is less than 16 dB. Fig. 2-3 shows us that this characteristic is consistent at different azimuth angles, while the magnitudes differ. The Matlab code of HRTF data plotting is given in Appendix I.

Fig. 2-2 The average HRTF in 0°-180° azimuth angle range
Fig. 2-3 Mesh plot of HRTF Measurements

The middle ear consists of the tympanic membrane or eardrum and a set of small bones called ossicles: the malleus (hammer), incus (anvil), and stapes (stirrup). The middle ear is connected to the throat through the Eustachian tube.

The middle ear has the duty of impedance matching. Both outer and middle ear is filled with air, whereas inner ear is filled with liquid. The air has low mechanical impedance, which means that it has low acoustic pressure and high particle velocity resulting from low density and high compressibility [13]. On the other hand liquid has high mechanical impedance. Because of the difference in mechanical impedance most of the sound is reflected at an air/liquid interface. The area difference between the eardrum and oval
window at cochlea is the first source of compensation. The eardrum has an approximate area of 60 mm², while the oval window, which is transmitting the sound into the liquid, has an approximate area of 4 mm². This difference in the area increases the sound wave pressure 15 times. The vibration of the eardrum is sensed by the hammer and through the action of other bones the sound pulse is amplified, which is the second source of compensation for the impedance difference.

The atmosphere exerts much more pressure on the eardrum than any other sound, and if there were no compensatory mechanism it would rupture. The Eustachian tube equalizes the atmospheric pressure on both sides of the eardrum. If this tube were open permanently we would not be able to hear anything, because it would be balancing also the sound pressure. Therefore it operates only during the action of swallowing [14].

The middle ear has also a safety mechanism for high intensity sounds. For high intensities, the muscles controlling the motion of the ossicles change their tension to reduce the amplitude of the motion of the stapes, thereby protecting the delicate mechanism of the inner ear. This process is called as acoustic reflex [10]. The weakness of the protection is the fact that it takes approximately 0.5 ms after the first perception of the loud sound to activate the reflex. Therefore it offers no protection for sudden impulsive sounds such as gunshots.

2.1.2 Inner Ear

It consists of three parts: the vestibule (entrance chamber), the semicircular canals (organs of balance), and the cochlea. Middle ear is connected to inner ear through the vestibule at two openings, the oval window and the round window. Oval window is
sealed with stapes and its support and the round window is sealed with a thin membrane, so that there isn’t any liquid leakage from the inner ear. Except these two openings the rest of the inner ear is surrounded by bone. The semicircular canals don’t play any role in hearing but in balancing.

The cochlea is a very critical organ in our hearing. Its name is derived from a Greek word for snail. It has roughly circular cross section, making 2.5 turns and has an approximate length of 3.5 cm. The cross-sectional area decreases in irregular manner and the cochlea has a total volume of 0.05 cm$^3$. The cochlea is partitioned into two channels: the upper gallery (scala vestibuli) and lower gallery (scala tympani). These two galleries are connected two each other at the apex of the cochlea through a small opening (helicotrema). On their other ends upper gallery is connected to oval window and the lower gallery is connected to the round window.

In Fig. 2-4 [10] a cross-section of one of the turns of the cochlea is given. Besides the two galleries there is an isolated part in cochlea, called the cochlea duck, where the auditory nerves are located. This part is isolated from the galleries by the two cells thick Reissner’s membrane and Basilar membrane. The organ of corti, carrying the hair cells, is located on Basilar membrane. The bony ledge carries the auditory nerve and at the end of bony ledge the nerve fibers enter the basilar membrane. The tectorial membrane lies above the basilar membrane. It is attached at one end to the bony ledge and at the other end it projects into the cochlear liquid. The hairs from the hair cells in the organ of corti extend to the under surface of the tectorial membrane. The organ of corti has four rows of hair cells all along the cochlea. There are approximately 30000 hair cells, whose 3500 are located in the inner row, where there are less vulnerable to damage.
The upper and lower gallery of the cochlea is filled with perilymphatic fluid, whereas the cochlea duct is filled with endolymphatic fluid. When the ear mechanism is excited by sound the stapes creates fluid disturbance through oval window. This disturbance travels down the upper gallery toward the apex of the cochlea, through the helicotrema, into the lower gallery, and back to the round window, which acts as a pressure-release termination. Since the tectorial membrane is attached to the bony ledge and the organ of corti is attached to the basilar membrane, the vibration difference between them flex the hairs, thereby exciting the nerve endings attached to the hair cells into producing electrical impulses. The basilar membrane is stiffest near the oval window and becomes
more flexible toward the opposite end. This decreasing stiffness makes the membrane resonate frequency selective through the cochlea. When a high frequency signal hits the cochlea, the basilar membrane resonates where it is stiff, resulting in the excitation of nerve cells close to the oval window. Fig. 2-5 shows the functional diagram of the human ear [13], and Fig. 2-6 [15] shows the approximate response positions of cochlea for different frequencies. This frequency dependent excitation pattern is called the place principal. The nerve cells encode the audio information by producing an action potential, which is an electrical impulse, in response to each cycle of the vibration. This works up to 500 Hz, which is the maximum rate that the neurons can produce action potentials. This problem is solved by human ear by allowing several nerves to take turns performing a single task. For example a 2 kHz tone can be represented by 5 cells alternately firing 400 times per second. This is called the volley principle [13].

![Functional Diagram of Human Ear](image)

**Fig. 2-5 Functional Diagram of Human Ear**

![Place Principle of Cochlea](image)

**Fig. 2-6 Place Principle of Cochlea**
Like the middle ear also the inner ear has a protection mechanism. The brain controls the sensitivity of the inner ear by returning signals through the Efferent Nerve Fibers. These fibers use the outer hair cells in order to protect the inner hair cells. The outer hair cells restrict or permit vibrations to reach to inner hair cells. This way the cochlea is protected in noisy environments and becomes more sensitive in quite environments. This feedback system is called the *Efferent System* [16].

### 2.2 Limits of Human Hearing

The cooperation of the individual parts described in section 2.1 establishes our hearing. Both intensity and frequency ranges and loudness concept of human hearing are explained in this section.

#### 2.2.1 Dynamic Range

The dynamic range of human hearing is defined as the difference between the audible weakest sound and the upper comfortable level, where discomfort starts. The nominal threshold of hearing ($P_0$) at 1 kHz is measured as $2 \cdot 10^{-5}$ Pa. In order to derive the intensity of this pressure, the acoustic resistance should be defined. The acoustic resistance $'r'$ varies with temperature because of its dependence on density of air $'\rho'$ and speed of sound $'c'$ according to Eq. (2-2). At 0 °C the acoustic resistance is 428 Rayls.

$$r = \rho \cdot c \quad (2-2)$$

whereas at room temperature it is reduced to 415 Rayls. To achieve a standard, the intensity threshold ($I_0$) is derived at room temperature:

$$I_0 = \frac{(P_0)^2}{r} = \frac{(2 \cdot 10^{-5})^2}{415} = 0.964 \cdot 10^{-12} = 10^{-12} \text{Watts/m}^2 \quad (2-3)$$
$10^{-12}$ Watts/m² is defined as the weakest audible sound intensity level at 1 kHz. The intensity or pressure of a sound signal is defined as the ratio of its value to these threshold values. The sound pressure level (SPL) and sound intensity level (IL) equations are:

$$SPL = 20 \cdot \log_{10}(P/P_0) \quad IL = 10 \cdot \log_{10}(I/I_0)$$

The dynamic range of human hearing is accepted as from 0 dB SPL ($2 \cdot 10^{-5}$ Pa) to 120 dB SPL (20 Pa). Over 120 dB SPL sound causes discomfort and damage in our ear. Table 2-1 shows some example sounds with their SPL and intensity values [13].

<table>
<thead>
<tr>
<th>Watts/cm²</th>
<th>dBSPL</th>
<th>Example Sound</th>
</tr>
</thead>
<tbody>
<tr>
<td>$10^{-2}$</td>
<td>140</td>
<td>Pain</td>
</tr>
<tr>
<td>$10^{-3}$</td>
<td>130</td>
<td></td>
</tr>
<tr>
<td>$10^{-4}$</td>
<td>120</td>
<td>Discomfort</td>
</tr>
<tr>
<td>$10^{-5}$</td>
<td>110</td>
<td>Jack hammers and rock concerts</td>
</tr>
<tr>
<td>$10^{-6}$</td>
<td>100</td>
<td></td>
</tr>
<tr>
<td>$10^{-7}$</td>
<td>90</td>
<td>OSHA limit for industrial noise</td>
</tr>
<tr>
<td>$10^{-8}$</td>
<td>80</td>
<td></td>
</tr>
<tr>
<td>$10^{-9}$</td>
<td>70</td>
<td></td>
</tr>
<tr>
<td>$10^{-10}$</td>
<td>60</td>
<td>Normal conversation</td>
</tr>
<tr>
<td>$10^{-11}$</td>
<td>50</td>
<td></td>
</tr>
<tr>
<td>$10^{-12}$</td>
<td>40</td>
<td>Weakest audible at 100 Hz</td>
</tr>
<tr>
<td>$10^{-13}$</td>
<td>30</td>
<td></td>
</tr>
<tr>
<td>$10^{-14}$</td>
<td>20</td>
<td>Weakest audible at 10 kHz</td>
</tr>
<tr>
<td>$10^{-15}$</td>
<td>10</td>
<td></td>
</tr>
<tr>
<td>$10^{-16}$</td>
<td>0</td>
<td>Weakest audible at 3 kHz</td>
</tr>
<tr>
<td>$10^{-17}$</td>
<td>-10</td>
<td></td>
</tr>
<tr>
<td>$10^{-18}$</td>
<td>-20</td>
<td></td>
</tr>
</tbody>
</table>

Table 2-1 Sound examples for dynamic range of human hearing

Table 2-1 shows us that human hearing is not only intensity dependant. The weakest audible sound intensity differs according to the frequency of that sound.
2.2.2 Frequency Range

The frequency range of human hearing is generally accepted to be 20 Hz to 20 kHz. As shown in Fig. 2-6 human cochlea splits the sound signal approximately into octave bands. The term octave means a factor of two in frequency. Human hearing covers approximately 10 octaves. The bands and their characteristics are given in Table 2-2 [17]:

<table>
<thead>
<tr>
<th>Octave</th>
<th>Frequency Range</th>
<th>Characteristics</th>
</tr>
</thead>
<tbody>
<tr>
<td>1&lt;sup&gt;st&lt;/sup&gt;</td>
<td>20-40 Hz</td>
<td>Low Bass - These frequencies add fullness, power and boom to sound. Lowest notes of bass, piano and tuba fall into this category.</td>
</tr>
<tr>
<td>2&lt;sup&gt;nd&lt;/sup&gt;</td>
<td>40-80 Hz</td>
<td>Upper Bass - These frequencies provide a balance in the structure of sound. Without them, sound is thin. The lower tones of the cello, trombone and rhythm sections produce sounds in this range.</td>
</tr>
<tr>
<td>3&lt;sup&gt;rd&lt;/sup&gt;</td>
<td>80-160 Hz</td>
<td>Midrange - Sounds get their intensity from this range of frequencies. Fundamentals and lower harmonics of most sound sources fall into this category.</td>
</tr>
<tr>
<td>4&lt;sup&gt;th&lt;/sup&gt;</td>
<td>160-320 Hz</td>
<td>Upper Midrange - Humans hear this range of frequencies best. 3,000-3,500 Hz contains information, which improves the intelligibility of speech and lyrics. If this band is incorrectly processed, sound becomes unpleasant. Frequencies above 3,500 Hz give sound realism and clarity. Listeners perceive sound in this section of the octave (and up to about 6,000 Hz in the 9&lt;sup&gt;th&lt;/sup&gt; octave) as being close. Thus, 3,500 - 6,000 Hz is known as the presence range.</td>
</tr>
<tr>
<td>5&lt;sup&gt;th&lt;/sup&gt;</td>
<td>320-640 Hz</td>
<td>Treble - Frequencies in this range give sound sparkle and brilliance. Most humans do not hear much beyond 16,000 Hz.</td>
</tr>
<tr>
<td>6&lt;sup&gt;th&lt;/sup&gt;</td>
<td>640-1,280 Hz</td>
<td></td>
</tr>
<tr>
<td>7&lt;sup&gt;th&lt;/sup&gt;</td>
<td>1,280-2,560 Hz</td>
<td></td>
</tr>
<tr>
<td>8&lt;sup&gt;th&lt;/sup&gt;</td>
<td>2,560-5,120 Hz</td>
<td></td>
</tr>
<tr>
<td>9&lt;sup&gt;th&lt;/sup&gt;</td>
<td>5,120-10,240 Hz</td>
<td></td>
</tr>
<tr>
<td>10&lt;sup&gt;th&lt;/sup&gt;</td>
<td>10,240-20,480 Hz</td>
<td></td>
</tr>
</tbody>
</table>

Table 2-2 Octave bands of human hearing and their characteristics
As shown in Table 2-2 each octave gives different characteristics to the sound signal. Because of the logarithmic frequency distribution of human hearing, the bandwidths of octave bands are increasing along the frequency axis. This shows that different, but equal amount of audio information is located between 20 Hz - 40 Hz and 10 kHz – 20 kHz. Therefore if we omit the 10th octave of the sound signal, the signal will still carry 90% information.

The male speech lies between 100 Hz and 8 kHz, whereas the harmonics of female speech can reach up to 10 kHz. The frequency and dynamic ranges of speech and music are plotted in Fig. 2-7 [18]. The solid line shows the border of normal hearing. We observe that human speech can be totally extracted if the original signal is low-pass filtered up to 8 kHz. On the other hand music covers a wider frequency band than the speech does.

![Diagram](image-url)  
**Fig. 2-7** Frequency and intensity ranges of speech and music
2.2.3 Loudness

As the border of hearing in Fig. 2-7 shows, the loudness of an equal intensity sound can vary drastically at different frequencies. The loudness is a psychophysical quantity and it depends on intensity, frequency as well as our neural system. Therefore it can only be measured by a human listener. The most commonly used loudness data is the Fletcher & Munson curves [14]. The measurements are done in the age group of 18 to 25 years with normal hearing. As reference 1 kHz tone is taken and the listener is asked to bring the signal as loud as it was at 1 kHz throughout the frequency sweep. Each time the intensity of the 1 kHz sound is increased by 10 dB and the same measurements are performed through the whole dynamic range of hearing. The unit of perceived loudness is the “phon”, which takes the SPL level of 1 kHz signal as reference. That means signals, which are on the same contour with a 10 dB SPL 1 kHz signal are at 10-phon loudness level. Fig. 2-8 shows Fletcher&Munson equal loudness contours [19].

![Fig. 2-8 Fletcher & Munson equal loudness contours](image)
From Fig. 2-8 we observe the high sensitivity of human ear for signals between 1 kHz and 6 kHz. Listeners can detect sounds as low as 0 dB SPL at 1 kHz, but require almost 40 dB SPL at 100 Hz. Listeners can tell that two tones are different if their frequencies differ by more than about 0.3 % at 1 kHz. At 100 Hz this increases to 3 %.

### 2.3 Hearing Loss Types

The intricate structure of human ear can be damaged by different sources. The degrees of hearing losses and their effects are given in Table 2-3. Hearing Loss is generally separated into two categories: ‘Conductive Hearing Loss’, ‘Sensorineural Hearing Loss’.

<table>
<thead>
<tr>
<th>LOSS</th>
<th>CLASSIFICATION</th>
<th>EFFECTS</th>
</tr>
</thead>
<tbody>
<tr>
<td>0-15 dB SPL</td>
<td>Normal hearing</td>
<td>-</td>
</tr>
<tr>
<td>16-25 dB SPL</td>
<td>Borderline normal (children)</td>
<td>-</td>
</tr>
<tr>
<td>15-25 dB SPL</td>
<td>Slight</td>
<td>Minimal difficulty with soft speech</td>
</tr>
<tr>
<td>25-40 dB SPL</td>
<td>Mild</td>
<td>Difficulty with soft speech</td>
</tr>
<tr>
<td>40-45 dB SPL</td>
<td>Moderate</td>
<td>Frequent difficulty with normal speech</td>
</tr>
<tr>
<td>46-70 dB SPL</td>
<td>Moderate-severe</td>
<td>Occasional difficulty with loud speech</td>
</tr>
<tr>
<td>71-90 dB SPL</td>
<td>Severe</td>
<td>Frequent difficulty with loud speech</td>
</tr>
<tr>
<td>&gt; 91</td>
<td>Profound</td>
<td>Near total loss of hearing</td>
</tr>
</tbody>
</table>

Table 2-3 Hearing loss degrees and their effects
2.3.1 Conductive Hearing Loss

Conductive hearing loss is caused by damage to, or a malfunction of the outer and middle ear. That means any hearing loss caused by a problem on the path up to inner ear is called the conductive hearing loss. Some of the disorders, which cause conductive hearing loss, can be listed as:

- Wax: Wax can build up and block sound from passing through the auditory canal.
- Otosclerosis: It causes immobilization of middle ear bones. It can be caused by deposits forming between, or dislocation of the ossicles.
- Malformation: The malformation in the ear canal can cause hearing loss.
- Rupture: The rupture of the ear drum, which can be caused by excessive air pressure or physical contact.
- Otitis media: Middle ear infection that causes fluid formation on the middle ear lining.
- Cholesteatoma: Tumor growth on eardrum.

The characteristics of the conductive hearing loss can be observed at an audiogram. The audiogram is extracted through presentation of pure tones into impaired persons each ears through earphones. The test is done at 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz and 8 kHz and the hearing loss at these frequencies are recorded. Some audiologists include 6 kHz too [20]. Besides the air conduction measurements also bone conduction audiogram is extracted by placing an oscillator behind the patients ear at ‘mastoid process’ and vibrating the bony structure of the skull, which brings the cochlea into vibration. This way the outer and middle ears are by-passed and the cochlea is tested for disorder. A typical conductive hearing loss audiogram is given in Fig. 2-9 [21].
The audiogram example for conductive hearing loss in Fig. 2-9 shows that, even though the air conduction screening shows a hearing loss, the bone conduction test doesn't show any impairment. It is because of the fact that the inner ear is functioning properly. Conductive hearing losses can be treatable either medically or surgically.

2.3.2 Sensorineural Hearing Loss

This type of hearing loss covers the disorder in the inner ear and in auditory neural system. Commonly the problem is the damage of the haircells inside the cochlea. The hearing can diminish gradually or suddenly as a trauma. The result is permanent hearing loss, which can not be cured with medicine or surgery. The need of hearing instruments is mainly for this type of hearing loss. Fig 2-10 shows a typical audiogram of sensorineural hearing loss, which shows an increasing loss at high frequencies. We observe that, both bone conduction and air conduction screening gives the same result.

2.3.2.1 Causes of sensorineural hearing loss and their characteristics
Aging: All human beings suffer a loss of hearing as part of their aging process. This sensorineural hearing loss is not associated with environmental conditions, but the changes in body chemistry. Table 2-4 shows the effect of aging on both male and female hearing. We observe that aging has more effect on high frequencies than low frequencies.

<table>
<thead>
<tr>
<th>Age</th>
<th>Frequency</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>500 Hz</td>
</tr>
<tr>
<td></td>
<td>Male (dB)</td>
</tr>
<tr>
<td>25</td>
<td>0</td>
</tr>
<tr>
<td>30</td>
<td>0</td>
</tr>
<tr>
<td>40</td>
<td>1</td>
</tr>
<tr>
<td>50</td>
<td>3</td>
</tr>
<tr>
<td>60</td>
<td>6</td>
</tr>
</tbody>
</table>

Table 2-4 Sensorineural hearing loss with age
It also shows us that aging causes more hearing loss for male than for female [11].

**Acoustic Trauma:** Short duration sound of sufficient intensity, like gunshot or explosion, may result in an immediate, severe, and permanent hearing loss, which is termed acoustic trauma. Depending on the intensity of the sound the organ of corti can be torn apart.

**Noise induced hearing loss:** Repeated exposure to sound at moderate intensity can cause permanent threshold shift of hearing. It is typically seen in industry, where the work environment has high noise pollution. As the Fig. 2-11 shows this sensorineural hearing loss has a dip around 4 kHz. 250 workers had been exposed to sound levels of approximately 100 dB SPL [22]. It shows a combination of aging and noise effect.

![Graph showing noise induced threshold shift](image)

*Fig. 2-11 Noise induced threshold shift*
**Ototoxicity:** It is the ear poisoning caused by drugs or chemicals. Some antibiotics, anti-cancer drugs and excessive aspirin can cause this sensorineural loss. They damage the hair cells or the auditory nerves [23]. The hearing loss typically starts at high frequencies and progresses into speech understanding range. It shows a ski-slope high frequency loss as plotted in Fig. 2-12 [24].

![Hearing Loss Graph](image)

**Fig. 2-12 Ototoxicity hearing loss**

**Meniere’s Disease:** It is caused by the change in fluid volume inside the cochlea. This effects the hair cells. An audiogram showing its effect is plotted in Fig. 2-13 [24]. The audiogram shows low frequency loss.

**Injuries:** Head injuries, which cause a reduced blood supply, can effect the cochlea.

**Fever:** High fevers for a prolonged time can harm inner ear structure.

**Other diseases:** Meningitis or tumor growth in auditory nerve can cause hearing loss.
2.3.2.2 Dynamic range reduction and loudness growth

The sensorineural hearing loss shifts the hearing threshold level. On the other hand it doesn’t effect the upper comfortable level of hearing. This reduces the dynamic range of hearing to a narrower range. This reduced dynamic range changes the loudness growth in the ear. That means in Fig. 2-8 even though the high sone contours aren’t displaced the lower phon contours shift up and the contours get closer. Therefore the growth of perceived loudness with increasing sound intensity is abnormally rapid. This growth is shown for mid-frequency range in Fig. 2-14. The dashed line shows the loudness growth for impaired hearing and solid line shows the loudness growth for healthy hearing. As it can be seen after 100 dB SPL the loudness of the signal is equal to both the normal person and the hearing impaired person. The impaired person has a hearing loss of 40 dB, where the impaired loudness growth curve starts.
2.4 Discussion on hearing instrument design constraints

In previous sections the anatomy of human ear and its characteristics are clarified. Different type of hearing losses and their signatures are pointed out. In this section the impact of these characteristics on hearing instrument design are pointed out.

**Auditory canal compensation:** It is shown that the outer and middle ears are the transmission parts of the human hearing. The resonance characteristic of the ear canal is important for hearing instrument design, because of the fact that the resonance frequencies lie in the frequency range of human speech, which is the target signal for hearing aid. By placing the microphone deeper in the ear canal, the natural amplification of the ear canal contributes to the general efficiency of the hearing instrument. Smaller instruments require less power than larger hearing instruments to treat the same degree of
hearing loss. Fig. 2-15 is comparing the relative gain contributed through hearing instrument and the ear canal [25]. This example shows the aid for a slight hearing loss. Using the resonance characteristics of the ear canal the gain difference between completely in the canal and in the ear hearing aids are compensated.

Fig. 2-15 Auditory canal compensation

**Limits of amplification:** Both middle ear and inner ear has protection mechanisms, but they don't protect against short time high intensity sounds. Therefore in hearing instrument design special care should be given to keep the sound signal in comfortable range of human hearing. That means the sound intensity shouldn't exceed 120 dB SPL.

**Frequency resolution:** 80% of the patients who are suffering from hearing loss has sensorineural loss, which can not be cured. Therefore hearing instrument designers are focused on this portion of hearing loss. As plotted and tabulated in section 2.3.2 sensorineural hearing loss can have different type of frequency characteristics. It is because of the fact that the cochlea is operating as a filterbank and if by any cause the
part of the organ of corti sensing a specific band is damaged, a narrowband hearing loss occurs. Therefore the hearing instrument should have a high frequency resolution in order to compensate properly. Considering that the bandwidth of a telephone line is between 200 Hz and 3.2 kHz, a frequency range up to 8 kHz would cover almost the whole speech components and most of the music components as demonstrated in Fig. 2-7. In terms of octave scale 90% of the sound information would be covered.

**Compression:** Because of the dynamic range reduction and rapid loudness growth linear amplification can not be applied to sound signals. It would amplify soft signals to audible range but loud signals to uncomfortable range. Therefore hearing instruments apply compressive amplification, which maps the sound signal into dynamic range of impaired person. It is demonstrated in Fig. 2-16 [26].

![Compressive amplification](image)

**Fig. 2-16 Compressive amplification**
**Processing Delay:** The time-difference between the sound entrance into microphone and sound exit from the loudspeaker is called the processing delay. As the bone conduction audiogram proves, we don't only hear through our ear canal. Even though our ear canal is totally blocked we can hear ourselves through the vibrations from our skull. There shouldn't be much time difference between the processed sound and original sound, so that we don't hear our echo. The processing delay or in other words group delay should be less than 10 ms and it should be fairly constant so that the original sound is not distorted.

**Power Consumption:** Another important point for hearing instrument design is the power consumption. The processing power should be kept as small as possible so that the battery life of the instrument is long. It is inconvenient to change the battery of the hearing instrument often.
Chapter 3

DYNAMEQ-II HEARING INSTRUMENT

This chapter covers the research on Gennum Corporation’s DynamEQ-II hybrid WDRC (Wide dynamic range compressor) system. The digital simulator of the analog hearing instrument is created. The limitations of the hearing instrument are discussed.

3.1 Signal representation

As it is pointed out in first chapter the signal coming from the microphone has to be represented digitally. This representation is done by converting the analog electrical signal into digital electrical signal, which is represented by 1’s and 0’s. This process is called the quantization. A very important point is the word length used for the analog signals representation. It is because of the fact that the data word length determines the dynamic range of the representation. Table 3-1 shows the dynamic range of different audio applications and devices [27]. The importance of the dynamic range coverage is observed by comparing the quality difference of an AM radio and CD player, which is partly dependent on sampling frequency too. The performance limitations of the hearing instrument microphone determines the necessary data word length. The lower limit of the microphone, which is called the noise floor, is the point at which the incoming signal can not be distinguished from the internally generated noise of the device. For a typical microphone it occurs at 23 dB SPL [28]. The upper limit is where the device gets into
saturation, which is approximately 110 dB SPL. That means a dynamic range of 87 dB need to be represented.

<table>
<thead>
<tr>
<th>Audio Device/Application</th>
<th>Dynamic Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>AM Radio</td>
<td>48 dB</td>
</tr>
<tr>
<td>Analog Broadcast TV</td>
<td>60 dB</td>
</tr>
<tr>
<td>FM Radio</td>
<td>70 dB</td>
</tr>
<tr>
<td>Analog Cassette Player</td>
<td>73 dB</td>
</tr>
<tr>
<td>Video Camcorder</td>
<td>75 dB</td>
</tr>
<tr>
<td>ADI Sound Port Codecs</td>
<td>80 dB</td>
</tr>
<tr>
<td>16-bit Audio Converters</td>
<td>90 to 95 dB</td>
</tr>
<tr>
<td>Digital Broadcast TV</td>
<td>85 dB</td>
</tr>
<tr>
<td>Mini-Disk Player</td>
<td>90 dB</td>
</tr>
<tr>
<td>CD Player</td>
<td>92 to 96 dB</td>
</tr>
<tr>
<td>18-bit Audio Converters</td>
<td>104 dB</td>
</tr>
<tr>
<td>Digital Audio Tape (DAT)</td>
<td>110 dB</td>
</tr>
<tr>
<td>20-bit Audio Converters</td>
<td>110 dB</td>
</tr>
<tr>
<td>24-bit Audio Converters</td>
<td>110 to 120 dB</td>
</tr>
</tbody>
</table>

Table 3-1 Dynamic Range Comparisons

Adding each bit to the word length means adding two more stages to the data representation levels. That means there will be an increase of $20 \cdot \log_{10}(2) = 6dB$ for Analog to digital converter (A/D). That shows why a 16 bit A/D covers 90 to 95 dB dynamic range as listed in the Table 3-1. It is shown that to achieve a high quality sound at least 16 bit representation is needed [28]. For simulations in Matlab 16 bit wav sound files are used. It is consistent with the word length used in a real hearing instrument. The biggest magnitude ‘1’ of the wav data is considered as 110 dB SPL.
3.2 DynamEQ-II Specifications

The dynamic range compression system has the conceptual model shown in Fig. 3-1 [29] and the functional block diagram is given in Fig. 3-2 [30]. The signal is separated into two channels with the aid of Linkwitz-Reilly state-variable filters. The filter outputs are applied to compressors, which map the input signal into comfortable hearing range of the patient. The two channels are added together without causing any signal distortion. It is because of the identical phase responses of filters. Additional gain can be applied before the signal is sent to the receiver.

![DynamEQ-II system model](image)

Fig. 3-1 DynamEQ-II system model

The Linkwitz-Reilly state-variable filter has tunable crossover frequency, which means that the cutoff frequency of the low-pass and high-pass filters can be shifted according to the crucial hearing loss frequency point.

3.2.1 Twin average Detection

Hearing instruments apply level dependent amplification to achieve compression. AGC (Automatic Gain Control) systems differ the way they measure the intensity level of the input sound. There are two major types: Peak detectors and average detectors [31].
peak detector systems the gain of the system is adjusted according to the peaks of the signal being sensed. On the other hand the average detector

Fig. 3-2 DynamEQ-II functional block diagram

systems adjust their gains according to the average value of the input sound over a definite time. This average is sensed by a low-pass filter. DynamEQ-II has average detector system.

Another important point about level dependent amplification is how fast the system responds to the change in the sound intensity. As it is indicated in second chapter, human ear doesn’t have a protection system for impulsive high intensity sound. Therefore the hearing instrument should respond very fast for high intensity changes in the signal. Otherwise the loud sound will be treated as a soft sound and it will be amplified to

34
uncomfortable levels. On the other hand the sound signal should hold its original envelope as much as possible so that the important speech clues are not lost [32].

This dilemma is solved in DynamEQ with twin average detection system. Two different averages of the sound signal are taken at the same time. The first low pass filter has an R-C time constant of 220 ms, which is the slow average detection. The other low pass filter has an R-C constant of 10 ms, which is the fast average detector. The fast average detector takes control, when its value exceeds the slow average detector by 6 dB. Otherwise the compression system is run by the value of the slow average detector [33]. The block diagram of these two detectors running the 2:1 compressor can be seen in Fig. 3-2 at left upper corner.

**Attack and Release Times:** Because of the dilemma of keeping the envelope characteristics and protecting against high intensity sounds a standard is defined for hearing instruments. This standard defines attack and release times, which show how quick the instrument reacts against a high intensity sound and how much time it takes for it to start applying a stable gain after an intensity reduction. The ANSI S3.22-1987 standard is as follows [33]:

*The attack time is defined as the time between the abrupt increase from 55 to 80 dB and the point where the level has stabilized to within 2 dB of the steady value for the 80 dB input sound pressure level. The release time is defined as the interval between the abrupt drop from 80 to 55 dB SPL and the point where the signal has stabilized to within 2 dB of the steady state value for the 55 dB SPL input sound pressure level.*
The theoretical envelope forms for DynamEQ system are given in Fig. 3-3. $V_{FAD}$ shows the voltage level at the output of the fast average detector and $V_{SAD}$ shows the voltage level at the output of the slow average detector. It is observed that the fast average detector quickly responds to the level change in the input sound signal and takes the control from the slow average detector. The peak at the edge of the transition from 55 dB SPL to 80 dB SPL shows the reaction time of the average detection system to the sudden sound intensity increase. When the sound intensity level drops from 80 dB SPL to 55 dB SPL the fast average detector shows again a quick response but this time the control is given to the slow average detector and it takes longer time to bring the output signal to the stable level. This is a desired response, because the gain is kept constant as much as possible, so that the important clues in the speech envelope are not changed.

Fig. 3-3 Demonstration of attack and release times for DynamEQ-II
3.2.2 Control Parameters

Besides the filter cutoff frequency adjustment DynamEQ-II has 5 control parameters. These parameters and their effect on I/O curve are explained below.

Compression Ratio: Both in high and low frequency channels independent from each other, compression ratios can be adjusted. The ratios can be between 1:1 and 1:4. The compression is applied to the sound between lower threshold level and upper threshold level, which is set as 95 dB SPL. The structure has a two-step compression adjustment. The first step is 2:1 compression before filtering and the second step is expanding or compression according to the desired overall compression ratio. These two blocks are synchronized. $R_{HI}$ and $R_{LO}$ are the resistors adjusting the compression ratios at high and low frequency channels. The effects of these two control parameters on I/O curve are plotted in Fig. 3-4 [29]. For this example the lower threshold level is set as 40 dB SPL.

![Output vs Input Graph](image)

Fig. 3-4 Compression ratio effect on I/O curve
**Lower threshold Control:** This parameter is adjusted through $R_{TH}$ resistor. It determines the lower threshold level of the I/O curve. Sound signals below this level are amplified linearly and the sound signals between this level and 95 dB SPL are amplified according to the compression ratio settings. The sound signals below this threshold are not fed into compressive amplification so that the low noise signals are not amplified and mask the important sound part. The effect of this control parameter, when all the other parameters are kept constant is plotted in Fig. 3-5 [29].

![Graph showing the lower threshold effect on I/O curve](image)

**Fig. 3-5 Lower threshold effect on I/O curve**

**MPO Control:** MPO (Maximum Power Output) adjusts the maximum output power of the instrument through $R_{MPO}$. It determines the ending point of the linear amplification above the upper threshold. Above MPO level the signals are pulled to this value. The effect of this control parameter is plotted in Fig. 3-6 [29].

**Gain Control:** This control applies gain to all signals below MPO level. It is a useful control parameter for patients who have both conductive and sensorineural hearing
loss. As indicated in second chapter conductive hearing loss has uniform frequency response and doesn’t cause loudness growth, because of the fact that it doesn’t effect the hair cells. It can be compensated by linear amplification. The effect of gain control on I/O curve of the system is plotted in Fig. 3-7.

Fig. 3-6 MPO effect on I/O curve

Fig. 3-7 Gain effect on I/O curve
3.3 DynamEQ-II Digital System Simulator

The analog DynamEQ-II dynamic range compression system is simulated digitally in Matlab's Simulink environment. The simulation is based on the specifications given in chapter 3.2. The overall view of the simulator is given in Fig. 3-8. The elements of the simulator and their specifications are explained below.

3.3.1 Simulator architecture

The simulator has the same system model given in Fig. 3-1. The 16-bit input sound data is split into two bands, where compressive amplification is applied. The simulator can be divided into three parts: Filters, average detectors and gain control units. The operations of these parts are explained below.

Fig. 3-8 DynamEQ-II digital system simulator

40
**Filters:**  The input sound signal is separated into low frequency and high frequency channels by the aid of the low pass and high pass filters. In digital filter design for hearing instruments the stop band attenuation should preferably cover the dynamic range of the input sound signal so that severe hearing losses can be compensated. On the other hand the limitations of the hearing instrument should also be taken into consideration. The phase response of the filters should be as linear as possible or identically nonlinear as it is the case in analog DynamEQ-II system. The phase linearity is necessary because of the fact that alternating group delay at different frequencies will cause distortion at the final output sound signal. The group delay should be less than 10 ms because of the self-hearing problem. In VLSI systems, multiplication is the most power consuming unit. Therefore the filters should be implemented with minimum number of multipliers. There is a trade-off between the filter attenuation and order of the filter, which determines the length of the group delay and number of arithmetic units. For this design, where there are only two channels, the dynamic range of the input signal can be covered without exceeding the group delay limitation.

Half-band filters are used to separate the input sound signal. A half-band filter is a symmetric finite-duration impulse response (FIR) digital filter, whose impulse response \( h_n \) has the following property:

\[
h_{2p} = 0 \quad \text{for} \quad p \neq 0
\]

(3.1)

It means that almost half of the coefficients of a half-band filter are zero except \( p=0 \), which corresponds to the mid coefficient of an even order filter. For a \( N = 4 \cdot M - 1 \) length filter it takes \( M + 1 \) multiplications and \( 2 \cdot M \) additions to compute each output.
These numbers are $2 \cdot M$ multiplications and $4 \cdot M - 2$ additions for an ordinary FIR filter. The constraint given in Eq. 3.1 requires the following relationship [34]:

$$H(e^{jw}) + H(e^{j(x-w)}) = 2 \cdot h_0$$  \hspace{1cm} (3.2)

In this addition the first term is the frequency response of the low pass filter. The second term is the frequency response of the complementary high pass filter, which is the reversed and shifted by $\pi$ response of the low pass filter. In order to get unity gain from this summation $h_0$ should be 0.5. This gives us the final form of the constraint required for an even order low pass filter in order to achieve total unity gain with its complement:

$$h(n) = \begin{cases} 
0 & n = odd \neq (N-1)/2 \\
0.5 & n = (N-1)/2
\end{cases}$$  \hspace{1cm} (3.3)

This low pass filter automatically satisfies the following conditions [34]:

$$f_p + f_s = 0.5 \hspace{0.5cm} ; \hspace{0.5cm} \delta_p = \delta_s$$  \hspace{1cm} (3.4)

$f_p$ is the normalized passband cutoff frequency whereas $f_s$ is the normalized stopband cutoff frequency. $\delta_p$ is the maximum passband error, whereas $\delta_s$ is the maximum stopband error. The low pass filter can be designed by McClellan-Parks algorithm, which is also known as remez algorithm [35]. The algorithm gives a half band lowpass filter when the conditions given in Eq. 3.4 are applied. The high pass filter can be extracted from the odd length (N) even order (N-1) low pass filter by a simple subtraction written both in frequency and time domain as follows:

$$H_h(z) = z^{-(N-1)/2} - H_l(z) ; \hspace{0.5cm} y_h(n) = y_l(n-(N-1)/2) - y(n)$$  \hspace{1cm} (3.5)
An example of a half band low pass filter and its complement is given below:

\[ h_l = [0 \ 0.2929 \ 0.5 \ 0.2929 \ 0] ; \ h_h = [0 \ -0.2929 \ 0.5 \ -0.2929 \ 0] \]  

(3.6)

The summation of these two impulse responses gives: \[ [0 \ 0 \ 1 \ 0 \ 0] \], which is nothing but the shift of the input signal by 2 samples. This proofs us that the perfect reconstruction is achieved with a half-band low pass filter and its complement high pass filter.

In simulator a 96\textsuperscript{th} order half band low pass filter is used. The magnitude and phase responses of the low pass filter and its complement, which is the high pass filter, are given in Fig. 3-9 and Fig. 3-10. The filters have 99 dB stop band attenuation and 0.0001 dB pass band ripple, which satisfies Eq. 3.4. At a sampling frequency of 16 kHz the cutoff frequencies are at 3.5 kHz and 4.5 kHz. The overlap of the filter responses is covering a range of 1 kHz. If the -3dB points of low pass and high pass filters are taken into consideration the overlap is 180 Hz.

![Frequency response graphs](image)

**Fig. 3-9** Frequency response of the simulator low pass filter

43
Fig. 3-10 Frequency response of simulator high pass filter

The group delay of the filters are 48 samples, which corresponds to 3ms delay at 16 kHz sampling frequency. The total magnitude response is unity gain, because of the half band characteristics explained above. If a buffer of 10 dB is left for noise floor, 90 dB gain can be applied in each channel.

**Average Detectors:** The twin average detection system is implemented. The analog DynamEQ-II utilizes RC circuits with time constants of 220 ms and 10 ms. These are lowpass filters monitoring a window of 220 ms and 10 ms and taking the averages in these windows. The digital counterpart of these systems are moving average detectors, which are digital lowpass filters. With 16 kHz sampling frequency, to take the average of 220 ms window 3520-tap and to take the average of 10 ms window 160-tap digital filters are needed. The outputs of the taps (delay elements) will be added together and divided to 3520 and 160. The frequency responses of these two filters are given in Fig. 3-11 and
Fig. 3-12. The slow average detector has a lower cutoff frequency and higher stopband attenuation than fast average detector. The same frequency response is obtained with a simpler structure given in Fig. 3-13. Instead of making 3520 additions at each cycle the structure adds the weighted new sample into average and subtracts the weighted sample, which is leaving the window from the average.

Fig. 3-11 Frequency response of slow average detector

Fig. 3-12 Frequency response of fast average detector
Fig. 3-13 Average detector structure in DynamEQ-II simulator

The average is stored in a memory element and the input signal is delayed 220 ms in order to subtract it from the average when that specific sample is leaving the window. The cost of this structure is one division and two additions at one cycle.

The average detectors monitor the average intensity of the sound signal because of the fact that the squared signal is fed into the detectors. After the average detection the signals should be converted into dB scale, so that the gain control unit can apply the control parameters. In dB conversion special attention should be given to zero valued samples, since the logarithm of zero gives minus infinity. The dB converter structure is given in Fig. 3-14. The intensity level is determined with the function:

\[ IL = 10 \cdot \log_{10}(x/10^{-11}) \]  \hspace{1cm} (3.7)

Fig. 3-14 dB converter in simulator
**Gain Control Unit:** The gain control unit takes the output from slow and fast average detectors and applies the four control parameters to achieve the gain response characteristics of the DynamEQ-II. The gain control block is given in Fig. 3-15. In order to achieve the same I/O curve with DynamEQ-II several relational and logical operations are applied. The effects of compression ratio, threshold, MPO and gain control parameters on gain control blocks I/O curve are simulated. Fig. 3-16 shows the effect of compression ratio parameter. The same effect given in section 3.2.2 is achieved. In simulation the lower threshold is kept constant at 20 dBSPL, the gain is kept at zero and MPO is kept at 100 dBSPL. The compression ratio is increased from 1:1 to 4:1 and the change is plotted. The same shifting effect is achieved. Lower threshold change is simulated by keeping the compression ratio at 2:1, the gain at zero and MPO at 100 dBSPL. Fig. 3-17 shows the same characteristics observed in section 3.2.2. The threshold level is increased from 20 dBSPL to 80 dBSPL with 20 dB increments. The effect of gain is also simulated. This time the lower threshold level is kept constant at 20 dBSPL, compression ratio is kept constant at 2:1 and MPO is kept at 100 dBSPL. The gain is
increased by 10 dB up to 40 dB. The simulation gives the same response as the gain effect in section 3.2.2. It is plotted in Fig. 3-18. Finally the MPO change is simulated. The threshold is kept constant at 20 dBSPL, the gain is kept constant at 0 dB and the compression is kept constant at 2:1. The MPO is decreased from 100 dBSPL to 80 dBSPL by a decrease of 5 dB. The same characteristic is achieved and plotted in Fig. 3-19.

Fig. 3-16 Simulated compression ratio change effect on I/O

Fig. 3-17 Simulated lower threshold change effect on I/O
The gain control unit determines the gain characteristic according to the parameters, whose characteristics are simulated above, and multiplies the original signal in each
channel with the necessary gain. The processed signals in two bands are added together and the final form of the output sound signal is achieved.

### 3.3.2 Attack and Release times of the simulator

As explained in section 3.2.1 attack and release times are important parameters for hearing instruments. They show the behavior of the device in time domain. Using the ANSI S3.22-1987 standard the attack and release time characteristics of the simulator are extracted. A sinusoidal signal at 2.5 kHz jumping from 55 dBSPL level to 80 dBSPL level is fed into gain control unit. The test signal is plotted in Fig. 3-20. It is showing the magnitude of the input signal. When its square is taken and the dB level is extracted, we observe that some signal go below 0 dB SPL. This is because of the fact that Simulink utilizes floating-point arithmetic. Even if a 16-bit signal is fed into system, when we take the square the signal starts getting represented with floating point. With 16-bit representation the part of the signal below 10 dBSPL would be represented as 10 dBSPL. The intensity level of this test signal is plotted in Fig. 3-21.

![Fig. 3-20 Magnitude of the input test signal](image)
Fig. 3-21 Intensity level of input test signal

The test signal is fed into simulator, whose threshold variable is 30 dB SPL, compression ratio is 1.5:1, gain is 0 dB and MPO is 110 dB SPL. The steady state output signal values for these control parameters are 79.4 dB SPL and 96 dB SPL respectively. In first simulation the gain unit switches the control over fast average detector as soon as it is higher than slow moving average by 6 dB. And it gives the control back to slow average as soon as fast average falls below the slow average plus 6 dB border. This uncorrelated control mechanism brings an additional peak at the transition from 55 dB SPL to 80 dB SPL as plotted in Fig. 3-22. When the fast average falls below the slow average plus 6 dB border the slow average detector takes the control but at that time its value is not equal to fast average detector and this causes discontinuity in the gain. This discontinuity deviates the signal from its final stable point and causes a longer attack time. The attack time for this simulation is 53.6 ms. When we check the characteristic of the fast average detector filter, we see that the filter operates over a window of 10 ms. The attack time is
more than 5 folds bigger than what it is suppose to be. The release time is 218 ms, which is a normal value comparing with the slow average detection window length of 220 ms. This attack time problem is solved by changing the control algorithm. Both the slow moving average ($V_s$) and fast moving average ($V_f$) are taken into consideration in determining the control average ($V_c$), if the fast moving average is bigger than slow moving average. The weighting parameter ‘$w$’ determines the effect of slow moving average when the fast moving average is bigger than the slow moving average. The response of the simulator to the test signal with gain control unit modified according to the Eq. 3.8 is plotted in Fig. 3-23.

$$V_c = \frac{(V_f - V_s) \cdot V_f + w \cdot V_s}{(V_f - V_s) + w}$$  \hspace{1cm} (3.8)

The control parameters are kept the same as the previous simulation. The weighting parameter is set to 6. It can be seen that the additional peak at the attack time is eliminated. The attack time is 6.4 ms and the release time is the same as before, 0.22 second.

![Fig. 3-22 Simulator response to test signal](image-url)
Fig. 3-23 Modified simulator response to test signal

Comparing the Figures 3-21 and 3-23 we observe that the signal, whose main energy lies between 20 dBSPL and 80 dBSPL is mapped to a signal, which lies between 40 dBSPL and 95 dBSPL. Because of the relaxed release time response the lower level of the main lobe of the mapped signal is shifted from 50 dBSPL range to 40 dBSPL range. Therefore the effect of 1.5:1 compression can not be observed well.

3.3.3 DynamEQ-II simulator response to speech signal

For hearing instruments the target signal is the speech. The response of the DynamEQ-II simulator to a speech signal is observed. The simulator is run for a hearing loss of 25 dB at frequencies higher than 4 kHz and no loss for frequencies lower than 4 kHz. The input signal is a female speech: ‘A lion was awaken from its sleep’. The time domain waveform of the input signal is given in Fig. 3-24. The histogram showing the intensity levels of the input signal is given in Fig. 3-25. It is observed that most of the energy of the signal is lying between 50 dBSPL and 85 dBSPL, which is a typical range for speech.
Fig. 3-24 Input speech waveform

Fig. 3-25 Histogram of the intensity levels of input speech

Since there isn’t any loss at low frequencies the gain control unit in low-channel keeps the signal as it is. At high channel the I/O response is adjusted for the compensation of hearing loss of 25 dB. This is achieved by applying 2:1 compression ratio starting from the threshold of 40 dB SPL. The gain parameter is kept zero and MPO is set to 110
dBSPL. The high channel I/O curve is plotted in Fig. 3-26. The signal is mapped to 25-110 dBSPL range and special care is given not to emphasize the input signals below 40 dBSPL.

![Image of High Freq Channel I/O Curve]

Fig. 3-26 High channel I/O curve for speech simulation

The output waveform of the simulator is plotted in Fig. 3-27. We observe amplitude increase and slight changes in waveform envelope.

![Image of Simulator Output Waveform]

Fig. 3-27 Simulator output waveform
The histogram of the intensity levels at the simulator output is given in Fig. 3-28. The upper limit of the intensity level is kept almost the same, but there are more samples lying close to upper limit. The important point is that this histogram is showing the total response of the low channel and high channel. Since the signal in the low channel is kept as it is we can't make healthy observation in time domain.

![Simulator output histogram](image)

Fig. 3-28 Simulator output histogram

The spectrum of the input speech and output speech are plotted and the effect of the hearing instrument is observed. Fig. 3-29 shows the spectrum of the input signal. It is observed that most of the energy of the input signal is lying up to 3.5 kHz range. For healthy hearing the low intensity level of the signal above this frequency range is enough to extract the information out of it. Fig. 3-30 shows the spectrum of the output signal. The response of the hearing instrument can be observed very well. The part of the signal over 4 kHz is emphasized for impaired hearing, but the rest is kept as it is.
3.4 Discussion

In this chapter the state of the art analog hearing instrument DynamEQ-II is simulated using digital signal processing structures. Except the frequency shift property of the hearing instrument, the rest of its characteristics are simulated successfully. The response
of the instrument to speech signal for the high frequency hearing loss of 25 dB case is
demonstrated.

The success of the instrument is subjective. As indicated in second chapter the perception
of loudness is psychophysical. Even though research on generating a performance metric
has been done [37] [38], a standard metric can not be defined. Therefore to evaluate the
performance of the hearing instruments field tests are performed, where patients with
variety kind of hearing losses wear the instruments in daily life.

To evaluate the performance of the hearing instruments in Simulink environment
research on hearing loss simulators is done.

An important disadvantage of the DynamEQ-II system is the poor frequency resolution.
The moving cutoff frequency doesn’t give enough flexibility to compensate some hearing
losses, like the notch type noise induced hearing loss given in Fig. 2-11 or Meniere’s
disease hearing loss given in Fig. 2-13. If the cutoff were set at the frequency where the
notch starts, than the high frequency levels, which don’t show much hearing loss, would
be overemphasized by the instrument.

Therefore research on digital filter bank structures suitable for hearing instruments is
carried out.
Chapter 4

DIGITAL FILTER BANK DESIGN FOR HEARING INSTRUMENTS

Research on digital filter banks, which can fulfill the requirements of a hearing instrument, is carried out. This chapter covers four different types of filter bank designs: Interpolated FIR (IFIR), QMF (quadrature mirror filter bank), DFT modulated and wavelet filter banks. The improvements achieved in state of the art designs are pointed out. The performances, limitations and suitability for hearing instruments of these filter banks are discussed.

4.1 Design Constraints

Filter bank is one of the core parts of the digital hearing instrument. Its performance determines the frequency resolution of the instrument and the gain limits in each frequency band. Therefore extended research is carried out on this topic. Most of the constraints listed at the end of the second chapter for the hearing instrument design are also the constraints for the digital filter bank design. The impacts of these constraints on digital filter bank design are explained in this section.

*Frequency Range:* It is shown in 2nd chapter that the frequency range of human hearing is from 20 Hz to 20 kHz. Because of the octave band characteristic of the human
hearing, good quality sound can still be achieved with half the frequency range coverage. In filter bank designs 16 kHz is taken as sampling frequency. Because of the fact that spectrum coverage of a system is half of its sampling frequency [39], up to 8 kHz of the input sound signal is taken. This will cover the first nine octaves in Table 2.2.

Number of Channels: Another important constraint is the frequency resolution. The monitoring of the hearing loss is done through audiogram, which takes measurements at eight different frequencies. Therefore 8 channels is an acceptable resolution for hearing instruments. More resolution at lower frequencies is preferable because of the octave characteristic of human hearing.

Stopband Attenuation: The stopband attenuation in each channel determines the gain range of the hearing instrument. It is preferable to get as much attenuation as possible. The order of the filter is proportional to stopband and passband attenuation [40]. When the order of the filter increases, the group delay and implementation cost increases too. Therefore the tradeoff between these parameters should be well adjusted to achieve optimum design.

Group Delay: The group delay of the filter bank should be as constant as possible, so that there won’t be any destruction in sound signal. If the group delay is kept too high, than the impaired person starts hearing himself or the visual image and aural image don’t map on each other. Some publications recommend a group delay less than 5 ms [41] and some recommend group delay less than 12.5 ms [42]. Therefore in our filter bank designs we tried to keep the group delay as small as possible and 10 ms is determined as the border.
For a filter with frequency response $H(e^{jwT})$ the phase $\tau_p$ and group delay $\tau_g$ are defined as [43]:

$$\phi(w) = \arg H(e^{jwT}) \quad \tau_p = -\frac{\phi(w)}{w} \quad \tau_g = -\frac{d\phi(w)}{dw}$$  \quad (4.1)

Since group delay is the derivative of the phase, constant group delay implies linear phase. Infinite impulse response filters (IIR) can achieve high stopband attenuation with fewer coefficients than finite impulse response filters (FIR) [44]. But they suffer in phase response. Their nonlinear phase response causes deviations in group delay. For audio applications the nonlinearity in phase might be acceptable, if the group delay deviations are minor. FIR filters can be designed with symmetrical coefficients, which assure the linear phase response and constant group delay [44].

**Power Consumption:** The power consumption is one of the most important constraints in filter bank design for hearing instruments. To achieve long battery life the circuitry inside the instrument should consume minimum power. The most power consuming arithmetic unit in binary VLSI implementation is multiplication [44]. Therefore the number of multiplication should be kept as small as possible. Besides the multiplication per unit time (MPU), addition per unit time (APU) is another important parameter to estimate the power consumption of the filter bank design [44].

**Size:** The research on filter bank designs is for CIC model hearing instruments. Therefore the size of the filter bank structure should be kept as small as possible. This implies the use of minimum number of digital elements in filter bank structure, which also effects the power consumption.
4.2 IFIR Filter Bank Design

This filter bank design method employs linear phase interpolated FIR filters and their complement. Properties of these filters and filter bank design procedure are explained in this chapter. Research on improving the overall magnitude response of the filter bank is demonstrated.

4.2.1. Interpolated Finite Impulse Response (IFIR) Filters

This technique enables the implementation of computationally efficient FIR filters. Instead of designing a very high order FIR filter, which satisfies the desired ripple and cutoff frequency specifications, the frequency response is achieved by cascading two or more FIR filters [45]. The basic idea is to implement the filter as a cascade of two FIR sections, where one section generates the sparse set of impulse response values with every $L^{th}$ sample being nonzero, and the other section performs the unwanted passband suppression. These two sections can be defined as prefilter and image suppressor. The block diagram is given in Fig. 4-1.

![Fig. 4-1 Block diagram of IFIR technique](image)

The technique is called the interpolated FIR, because of the fact that the image processor block is applying an interpolation scheme to the impulse response samples of the prefilter. This interpolation extracts the desired narrow band filter. The interpolation is
not applied to the input data, but the filter coefficients. Therefore it is more convenient to call the second block in Fig. 4-1 as ‘image suppressor’ instead of ‘interpolator’ [44].

The first step in the design process is to determine the constraints of the narrow band filter. As a design example the stopband, passband cutoff frequencies and stopband, passband ripples at 16 kHz sampling frequency are set as:

\[ f_p = 250Hz \quad f_s = 750Hz \quad \delta_p = \delta_s = 0.01 \]  \hspace{1cm} (4.2)

The direct design of the FIR filter using the McClellan-Park algorithm [35] requires an order of 62. The magnitude response is given in Fig. 4-2.

![Lowpass filter magnitude](image)

Fig. 4-2 Lowpass filter magnitude designed in one step

The same filter response can be achieved by employing the IFIR technique. The first step is to determine the L, which is the relaxation factor. L is taken as 8 for this example. Using the McClellan-Park algorithm a low pass model filter is designed, whose cutoff frequencies are L times bigger than the desired filters:
\[ f_p = 2000\text{Hz} \quad f_s = 6000\text{Hz} \quad \delta_p = \delta_s = 0.01 \] (4.3)

This filter is 6\textsuperscript{th} order. Its design constraints satisfy the conditions of a half band filter, which are given in Eq.3.4. Therefore \( h_M(n) \) has only four nonzero coefficients, besides the middle coefficient, which is 0.5. Its magnitude response is given in Fig. 4-3.

\[ h_M(n) = [-0.0506 \ 0.0 \ 0.2951 \ 0.5 \ 0.2951 \ 0.0 \ -0.0506] \] (4.4)

The next step is to zero pad this impulse response with \( L-1 \) samples. That means seven zero valued samples are inserted between each sample according to the following equation:

\[ h_{MS}(n) = \begin{cases} h_M(n/L) & n = iL, i = 0, \pm 1, \pm 2, \ldots \\ 0 & \text{otherwise} \end{cases} \] (4.5)

The magnitude response of \( h_{MS}(n) \) is given in Fig. 4-4. We observe that the first lobe in this magnitude response obeys the constraints of the desired narrow band lowpass filter. Therefore if the rest of the images are suppressed, the desired filter will be obtained.

![Graph showing the magnitude response of the filter](image)

**Fig. 4-3 Model filter magnitude response**

64
The image suppression can be done in one step by a higher cutoff frequency lowpass filter or by cascade of couple FIR filters. Using cascade of three FIR filters as image suppressor the final magnitude response in Fig. 4-5 is obtained. It satisfies the design constraints given in Eq. 4.2.
Using its symmetry the cost of direct design can be reduced to 32 multiplications and 64 additions [35]. On the other hand the total cost of IFIR structure can be reduced to 11 multiplications and 14 additions. There is a remarkable saving in computation.

4.2.2 Filter Bank Structure

The example given in previous section was describing the process of extracting the first passband of the filter bank. Each lobe in Fig. 4-4 is a passband of the filter bank and can be extracted using different type of FIR filters as image suppressors. The magnitude response given in Fig. 4-4 doesn't cover the whole spectrum. The complement of this zero padded half band filter is taken with the subtraction given in Eq. 3.5. Even though the zero padded filter is not a half band filter anymore, the coefficients, which are nonzero, are still the coefficients of a half band filter. Therefore the complement of this filter can be taken and the sum of these two filters is unity. The complement of the filter in Fig. 4-4 is plotted in Fig. 4-6.

![Graph showing frequency response](image)

**Fig. 4-6 Complement of zero padded model filter**
The total 8-band filter bank structure is given in Fig. 4-7. The uppermost lobe of the filter in Fig. 4-4 is left out as transition band. Only the first four lobes of this filter and the four lobes of the filter in Fig. 4-6 are taken into filter bank. Each pathway in the structure establishes a channel of the filter bank. The magnitude responses at each point of the structure are also plotted in Fig. 4-7. Each block represents the filter and its complement. For example the block $H_1(z)$ has the response of this filter and its complement response is also achieved with the subtraction given in Eq. 3.5. Since only zero padded half band filters are employed throughout the structure, the complement of each filter can be taken.

The symmetry properties of the coefficients in the structure are given in Eq. 4.6 [41]. Every symmetric coefficient in a filter impulse response brings one multiplication. Therefore the filter bank can be implemented with 27 multiplications. 8 of them are the multiplication with 0.5, which can be implemented as a shift register in the hardware [46]. The total number of additions is 45. The cost of the total filter bank is less than the direct design of its first band given in Fig. 4.2.

\begin{align*}
h_1(0) &= h_1(48), h_1(16) = h_1(32), h_1(24) = 0.5 \\
h_2(0) &= h_2(24), h_2(8) = h_2(16), h_2(12) = 0.5 \\
h_3(0) &= h_3(28), h_3(4) = h_3(24), \\
h_3(8) &= h_3(20), h_3(12) = h_3(16), h_3(14) = 0.5 \\
h_4(0) &= h_4(12), h_4(4) = h_4(8), h_4(6) = 0.5 \\
h_5(0) &= h_5(10), h_5(2) = h_5(8), h_5(4) = h_5(6), h_5(5) = 0.5 \\
h_6(0) &= h_6(6), h_6(2) = h_6(4), h_6(5) = 0.5 \\
h_7(0) &= h_7(30), h_7(6) = h_7(24), h_7(12) = h_7(18), h_7(15) = 0.5 \\
h_8(0) &= h_8(2), h_8(1) = 0.5
\end{align*}
Fig. 4-7 IFIR Filter Bank Structure
The magnitude response of each passband is given in Fig. 4-8. The stopband attenuation of the filter bank, or noise floor of the filter bank is 38 dB and passband ripples deviate between 0.2 dB and -0.3 dB. The lower and upper passband and stopband cutoff frequencies of the channel responses are listed in Table 4-1.

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Cutoff Freq.(Hz)</td>
<td>Cutoff Freq.(Hz)</td>
<td>Cutoff Freq.(Hz)</td>
<td>Cutoff Freq.(Hz)</td>
</tr>
<tr>
<td>1&lt;sup&gt;st&lt;/sup&gt;</td>
<td>-</td>
<td>-</td>
<td>280</td>
<td>745</td>
</tr>
<tr>
<td>2&lt;sup&gt;nd&lt;/sup&gt;</td>
<td>250</td>
<td>720</td>
<td>1300</td>
<td>1745</td>
</tr>
<tr>
<td>3&lt;sup&gt;rd&lt;/sup&gt;</td>
<td>1260</td>
<td>1710</td>
<td>2300</td>
<td>2740</td>
</tr>
<tr>
<td>4&lt;sup&gt;th&lt;/sup&gt;</td>
<td>2260</td>
<td>2700</td>
<td>3290</td>
<td>3745</td>
</tr>
<tr>
<td>5&lt;sup&gt;th&lt;/sup&gt;</td>
<td>3255</td>
<td>3715</td>
<td>4285</td>
<td>4745</td>
</tr>
<tr>
<td>6&lt;sup&gt;th&lt;/sup&gt;</td>
<td>4255</td>
<td>4715</td>
<td>5300</td>
<td>5745</td>
</tr>
<tr>
<td>7&lt;sup&gt;th&lt;/sup&gt;</td>
<td>5260</td>
<td>5700</td>
<td>6290</td>
<td>6745</td>
</tr>
<tr>
<td>8&lt;sup&gt;th&lt;/sup&gt;</td>
<td>6255</td>
<td>6700</td>
<td>7280</td>
<td>7750</td>
</tr>
</tbody>
</table>

Table 4-1 Cutoff frequencies of IFIR filter channel responses

It is observed that the main lobe of the first passband is 280 Hz wide. The rest of the channels have an approximate width of 580 Hz. It is a uniform filter bank except the first passband, which is half the width of the rest.

Only the phase response of the first passband is plotted in Fig. 4-9. As it can be seen it is linear. The rest of the channels have linear phase responses too.
Fig. 4-8 Magnitude responses of pass bands
Fig. 4.8 Magnitude responses of pass bands
The linear phase response of the passbands brings constant group delay. The group delay of the filter bank is 53 samples, which corresponds to 3.3 ms at 16 kHz sampling.

4.2.3 Optimization of delay characteristics of the filter bank

The total response of the filter bank is plotted in Fig. 4-10. Even though only complementary filters have been used in filter bank structure, the total response deviates from unity gain. It is because of the mismatch of phase characteristics of individual passbands. This destructive effect can be eliminated by the addition of delay elements to shorter channels [47]. The delay characteristics of the channels are given in Fig. 4-11. There are six shorter channels in the structures. Delay elements can be added to these paths without altering the magnitude responses of channels and changing the group delay of the filter bank. The problem is to find the optimum numbers of delay elements, which will be added to each channel, to obtain maximally flat frequency response.
Fig. 4-10 Total magnitude response of the filter bank

We observe ripple in the range of \(-11.4\) dB to \(+0.1\) dB. This will drastically alter the input signal.

Fig. 4-11 The delay characteristics of filter bank channels

The addition of delay elements is handled as an integer programming problem with six variables and six constraints [48]. The variables are the number of extra delay elements,
which will be added to the channels and constraints are the delay differences between the longest path of the structure and the shorter channels. The group delay shouldn’t be increased with these extra delay elements.

The addition of delay elements in the channel mainly affects the channels own passband. Therefore the optimization can be divided into six different optimization problems. The effect on the overall magnitude response at the end of each optimization process is plotted in Fig. 4-12.

The optimum result is plotted in Fig. 4-13 again. It is observed that the ripple is reduced to the range of +0.1 dB to −0.1 dB. The Matlab code of optimization process is given in Appendix II.

In Fig. 4-13 we can observe that the overall magnitude response doesn’t cover a range up to 8 kHz. It is because of the fact that the upper most lobe of the filter bank is left out as transition band.

![Graph showing magnitude response](image)

Fig. 4-13 Maximally flat overall magnitude response
Fig. 4.12 Optimalization Process of decay elements
4.2.4 The applicable gain for filter bank structure

The IFIR filter bank has a noise level of 38 dB. This noise level doesn’t mean that 38 dB of gain can be applied to a signal in the channel. The effect of gain increase in first channel of the filter bank is given in Fig. 4-14. The gain is increased from 0 dB to 35 dB by an increment of 5 dB and the effect of 38 dB gain is examined last. The responses are plotted all together. The impact on other channels is below 1 dB up to a gain of 25 dB. At 30 dB of gain it becomes 2 dB, at 35 dB of gain it becomes 5 dB and at 38 dB of gain it becomes 9 dB. It is observed that after 30 dB the increase of the gain in the channel brings additional increase of the gain in some adjacent bands.

The effect of gain increase is also observed in 5th channel and 7th channel. The overall magnitude response of the system with the gain increase in 5th channel is plotted in Fig. 4-15. The impact of channel 7 is plotted in Fig. 4-16. The same increase used for first channel is applied to these two channels.

Fig. 4-14 The effect of 1st channel gain increase on overall response
Fig. 4-15 The effect of 5th channel gain increase on overall response

Fig. 4-16 The effect of 7th channel gain increase on overall response

Almost the same characteristics observed for the first passband is observed for the 5th and 7th. It shows that we have to leave some noise buffer between the upper gain of the filter
bank and noise floor. This buffer should 8 dB for this filter bank design. 30 dB of gain effects the other bands 2 dB, which is an acceptable value.

4.2.5 IFIR Filter bank with -60 dB noise level

In section 4.2.4 it is shown that a filter bank with lower noise level than −38 dB is needed to compensate moderate level hearing loss. Therefore the design constraints of the filters are increased to 60 dB stopband attenuation. The structure of the filter bank is the same as in Fig. 4-7. The only difference is the increase in the order of filters to achieve higher stopband and passband ripples. In Fig. 4-17 all pass bands of the filter bank are plotted together. The passband ripples are between −0.02 dB and +0.02 dB. The noise floor is at −59 dB. It is shown in previous sections that the noise floor together is more important than the individual stopband characteristics. The effects of the individual stopband characteristics are observed best in gain increase effect on overall magnitude response. Therefore the passband magnitude responses are not plotted one by one.

![Magnitude responses of 60 dB IFIR design pass bands](image)

Fig. 4-17 Magnitude responses of 60 dB IFIR design pass bands
The lower and upper passband and stopband cutoff frequencies of the channel responses are listed in Table 4-2.

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Cutoff Freq.(Hz)</td>
<td>Cutoff Freq.(Hz)</td>
<td>Cutoff Freq.(Hz)</td>
<td>Cutoff Freq.(Hz)</td>
</tr>
<tr>
<td>1&lt;sup&gt;st&lt;/sup&gt;</td>
<td>-</td>
<td>-</td>
<td>240</td>
<td>765</td>
</tr>
<tr>
<td>2&lt;sup&gt;nd&lt;/sup&gt;</td>
<td>240</td>
<td>755</td>
<td>1245</td>
<td>1760</td>
</tr>
<tr>
<td>3&lt;sup&gt;rd&lt;/sup&gt;</td>
<td>1240</td>
<td>1760</td>
<td>2250</td>
<td>2760</td>
</tr>
<tr>
<td>4&lt;sup&gt;th&lt;/sup&gt;</td>
<td>2240</td>
<td>2750</td>
<td>3250</td>
<td>3760</td>
</tr>
<tr>
<td>5&lt;sup&gt;th&lt;/sup&gt;</td>
<td>3240</td>
<td>3760</td>
<td>4240</td>
<td>4760</td>
</tr>
<tr>
<td>6&lt;sup&gt;th&lt;/sup&gt;</td>
<td>4240</td>
<td>4750</td>
<td>5245</td>
<td>5760</td>
</tr>
<tr>
<td>7&lt;sup&gt;th&lt;/sup&gt;</td>
<td>5240</td>
<td>5750</td>
<td>6250</td>
<td>6760</td>
</tr>
<tr>
<td>8&lt;sup&gt;th&lt;/sup&gt;</td>
<td>6240</td>
<td>6750</td>
<td>7250</td>
<td>7760</td>
</tr>
</tbody>
</table>

Table 4-2 Cutoff frequencies of 60 dB IFIR filter bank

The width of the first pass band is 240 Hz and the rest is between 480 and 500 Hz. Comparing to Table 4-1, this design has more uniform response. On the other hand the width of the transition bands has been increased to 500 Hz. The difference between the Passband and stopband cutoff frequencies was approximately 450 Hz for the first design. This slight increase can bring more impact of the gain increase in the passband on the adjacent bands.

Total magnitude response of the filter bank is plotted in Fig. 4-18. Deviation from unity gain is in the range of +0.04 dB to −11.4 dB.
The same optimization process for overall magnitude response improvement is applied on this design too. The maximally flat overall magnitude response is plotted in Fig. 4-19. The ripples are reduced to the range +0.01 dB to -0.01 dB.

![Graph of overall magnitude response for 60 dB IFIR design](image1)

**Fig. 4-18 Overall magnitude response of 60 dB IFIR design**

![Graph of maximally flat magnitude response for 60 dB IFIR design](image2)

**Fig. 4-19 Maximally flat overall magnitude response for 60 dB IFIR design**
The effect of gain increase in the 7th channel is plotted in Fig. 4-20. In previous section we have seen that the gain increase response of different pass bands in IFIR structure are almost the same. Therefore only one channel is examined. The gain is increased by 5 dB up to 55 dB and the response for 59 dB gain is also plotted. We observe that for 59 dB gain the effect on the adjacent band becomes almost 20 dB. 55 dB gain brings a deviation of 7 dB in adjacent band. The influence of 50 dB gain on the adjacent band is below 3 dB, which is an acceptable value.

The buffer level between the noise level and gain should be again around 9 dB. Therefore gain up to 50 dB can be applied in this filter bank structure. It is enough to compensate moderate level hearing loss.

The cost of the 60 dB design is 42 multiplication and 75 additions. The additions also cover the cost of extracting the complementary filters. The group delay is increased to 103 samples, which corresponds to 6.4 ms at 16 kHz sampling frequency. It is below the 10 ms maximum group delay border.

Fig. 4-20 The effect of gain increase on 60 dB IFIR filter bank
4.3 QMF (Quadrature Mirror Filter) Bank Design

QMF is a multirate filter bank. A multirate filter bank is constructed from two parts: The analysis filter bank and the synthesis filter bank. The analysis filter bank filters the single input signal into multiple outputs and downsamples (subsamples) them by a factor N. The synthesis filter bank upsamples the signals by N and generates a single output by eliminating the aliasing [49].

The block diagram of a QMF bank is given in Fig. 4-21. It is a two-band filter bank structure, where \( H_0 \) and \( H_1 \) are analysis filters and \( G_0 \) and \( G_1 \) are synthesis filters. The upper limit of downsampling in a multirate filter bank is its number of channels [50]. Therefore a QMF bank can downsample only by 2.

![Fig. 4-21 Block diagram of QMF bank](image)

The design difficulty is to reconstruct the input signal at the filter bank output without distortion. That means \( y(n) \) should be equal to \( x(n) \). The multirate filter banks are used in subband coding and transmultiplexing for telecommunications [49]. In these applications the aim is to reduce the number of samples in channels. There isn’t much gain applied on signal.
In Fig. 4-21 \( H_0(z) \), \( G_0(z) \) are low pass filters and \( H_1(z) \), \( G_1(z) \) are high pass filters.

The operation of the filter bank can be described as follows: \( x(n) \) is separated into its low frequency \( v_0(n) \) and high frequency \( v_1(n) \) components. These signals are downsampled by 2, which has the effect of widening the frequency content of the signals by two [51]. \( z_0(n) \) and \( z_1(n) \) are upsampled by a factor of two, which shrinks the frequency content of the signal back to its original range, but adds an additional image of the signal into spectrum. These images are suppressed by \( G_0(z) \) in first channel and \( G_1(z) \) in second channel. An impulse is fed into system and responses of the system at different points are given in Fig. 4-22.

The effect of downsampling is expressed in time domain, frequency domain and \( z \) domain as [50]:

\[
z_k[n] = v_k[R \cdot n] ; \ Z_k(e^{j\omega}) = \frac{1}{R} \sum_{r=0}^{R-1} V_k(e^{j\frac{2\pi R}{R}}) ; \ Z_k(z) = \frac{1}{R} \sum_{r=0}^{R-1} V_k(W_k^r \cdot z^R) \quad (4.6)
\]

\( R \) is the downsampling ratio and \( W_k^r = e^{-j\frac{2\pi r}{R}} \). For two channel QMF bank \( R = 2 \) and \( W_2^1 = -1 \). Therefore \( Z_k(z) \) is written for \( k = 0,1 \) as:

\[
Z_k(z) = \frac{1}{2} \cdot [H_k(z^2) \cdot X(z^2) + H_k(-z^2) \cdot X(-z)] \quad (4.7)
\]

The effect of upsampling is defined in time domain, frequency domain and \( z \)-domain as:

\[
s_k(n) = \begin{cases} z_k(n/R) \quad n = 0, \pm R, \pm 2R, \ldots \\ 0 \quad \text{otherwise} \end{cases} ; \; S_k(e^{j\omega}) = Z_k(e^{jR\omega}) \; ; \; S_k(z) = Z_k(z^R) \quad (4.8)
\]

Therefore \( S_k(z) \) is defined for \( k = 0,1 \) as:
Fig. 4-22 Impulse responses of the first band of QMF bank
\[ S_k(z) = \frac{1}{2} \cdot (H_k(z) \cdot X(z) + H_k(-z) \cdot X(-z)) \quad (4.9) \]

After the synthesis filters the total response of QMF bank is written as:

\[ Y(z) = \frac{1}{2} \cdot G_0(z) \cdot [H_0(z) \cdot X(z) + H_0(-z) \cdot X(-z)] \]
\[ + \frac{1}{2} \cdot G_1(z) \cdot [H_1(z) \cdot X(z) + H_1(-z) \cdot X(-z)] \quad (4.10) \]

In order to achieve perfect reconstruction \( Y(z) \) should satisfy the following equation:

\[ Y(z) = z^{-N} \cdot X(z) \quad (4.11) \]

\( N \) is an arbitrary integer. That means the output should be a delayed version of the input.

The transfer function of the filter bank can be written in matrix form as follows:

\[ Y(z) = \frac{1}{2} \cdot [X(z) \quad X(-z)] \cdot \begin{bmatrix} H_0(z) & H_1(z) \\ H_0(-z) & H_1(-z) \end{bmatrix} \cdot \begin{bmatrix} G_0(z) \\ G_1(z) \end{bmatrix} \quad (4.12) \]

The matrix with \( H(z) \) is called alias cancellation matrix (AC Matrix). It has the analysis filter bank coefficients. The synthesis filter bank can be extracted from the defined analysis filter bank using the following equation [50]:

\[ \begin{bmatrix} H_0(z) & H_1(z) \\ H_0(-z) & H_1(-z) \end{bmatrix} \cdot \begin{bmatrix} G_0(z) \\ G_1(z) \end{bmatrix} = \begin{bmatrix} T(z) \\ 0 \end{bmatrix} \quad (4.13) \]

\( T(z) \) is the transfer function of the system. The zero term on the right side of the equation is to remove the aliasing. In order to achieve perfect reconstruction \( T(z) \) should satisfy Eq. 4.11, which means that it should be a delay. The synthesis filters can be written in terms of analysis filters and system transfer function as follows:
\[
\begin{bmatrix}
G_0(z) \\
G_1(z)
\end{bmatrix} = \frac{1}{H_0(z) \cdot H_1(-z) - H_0(-z) \cdot H_1(z)} \begin{bmatrix}
H_1(-z) & -H_0(z) \\
-H_0(-z) & H_0(z)
\end{bmatrix} \begin{bmatrix}
T(z) \\
0
\end{bmatrix}
\] (4.14)

In order to achieve an FIR solution for \(G(z)\)'s, the first term on the right hand side of the equation, which is the determinant of the AC matrix should be a constant delay.

The first solution to this problem was the quadrature mirror filter, which was defined with the relationships below:

\[
H_1(z) = H_0(-z); \quad G_0(z) = H_0(z); \quad G_1(z) = -H_0(-z)
\] (4.15)

When these equations are put into equation 4.13, it is seen that the zero aliasing condition is satisfied and the system transfer function \(T(z)\) becomes:

\[
T(z) = H_0(z) \cdot H_1(-z) - H_0(-z) \cdot H_1(z)
\] (4.16)

This transfer function can be made close to a delay, but optimization is needed.

Another solution is obtained from the observation that \(H_0(z) \cdot H_1(-z)\) is a lowpass filter and \(H_0(-z) \cdot H_1(z)\) is its complementary highpass filter [50]. It means that they are half band filters and their summation gives unit response. A half band low pass filter can be designed and through spectral factorization two lowpass filters can be extracted from this filter [52]. The half band filter is spectrally factorized such that \(H_1(-z) = H_0(z^{-1})\).

Through the following equations the filter bank can be built:

\[
G_0(z) = H_0(z^{-1}); \quad G_1(z) = H_0(-z); \quad H_1(z) = H_0(-z^{-1})
\] (4.17)

They automatically satisfy both the alias cancellation and transfer function constraints.
The Barnwell coefficients for 40 dB stopband attenuation filter bank is used to built the QMF bank [50]. The effect of gain increase in first channel is plotted in Fig. 4-23. The gain is increased by 5 dB up to 40 dB.

![Graph showing magnitude vs frequency](image)

Fig. 4-23 Gain increase effect on QMF banks overall response

It is observed that the gain should be kept below 35 dB for the QMF bank.

Our research is on 8-band filter bank, why did we explain a QMF bank? It is because of the fact that using the structure in Fig. 4-24, a QMF bank can be extended to an eight-band filter bank [53]. The analysis and synthesis filters are the same as the lowpass and highpass filters used in QMF bank. The signal is downsampled by 2 at each additional level in analysis filter bank. Between the analysis and synthesis filter bank the signal is downsampled by 8. At synthesis filter bank it is upsamled back to its original rate.

The signal processing is done between analysis and synthesis filter banks. The overall magnitude response of the filter bank gives unity response, since the same QMF properties are extended. But the impulse response of the system with gain applied between the analysis and synthesis filter banks doesn’t show the uniformity plotted for
the QMF bank in Fig. 4-23. 30 dB gain is applied to each channel, while the other channels are kept at 0 dB gain. The 8 different impulse responses of the filter bank are plotted together in Fig. 4-25. The individual channel magnitude responses have too much overlap.
The filter bank is useful for subband coding, which doesn’t apply much gain. The suitability of the filter bank for digital hearing aid is examined and the poor passband responses are demonstrated.

4.4 Wavelet filter bank

The theory and specifications of wavelets need another thesis to cover. The important points of the wavelet theory for filter bank design are pointed out and the generation of filter bank using wavelets is explained.

Wavelets are introduced in 1980’s as an alternative to Fourier transform. Wavelets are localized waves. Instead of oscillating forever they drop to zero.

The operation of wavelet transform can be described in discrete time as expression of input $x(n)$ in the wavelet basis as a set of coefficients $y_k(n)$ [54]. We can look at it as band pass filtering an input signal to obtain the coefficients. The coefficients are actually the signals in each channel and the filter coefficients $h_k(n)$ are the basis of the wavelet transform. The discrete wavelet transform can be expressed as [44]:

$$y_k(n) = \sum_{m=-\infty}^{\infty} x(m) \cdot h_k(2^{k+1} \cdot n - m), 0 \leq k \leq M - 2$$

$$y_{M-1}(n) = \sum_{m=-\infty}^{\infty} x(m) \cdot h_{M-1}(2^{M-1} \cdot n - m)$$

(4.18)

This equation can be implemented by using the dyadic analysis filter bank given in Fig. 4-26. It is providing octave band response. In this implementation $M = 6$ and $h_k$’s are cascade of two basis coefficients of an orthogonal wavelet set. This two coefficients give one decomposition low pass filter LP ($z$) and one decomposition high pass filter HP ($z$).
For example $H_0(z) = HP(z^2)$ and $H_1(z) = HP(z^2) \cdot LP(z^4)$. The rest of $H_k(z)$ is obtained by cascading the decomposition filters and downsamplers. The inverse discrete wavelet transform can be expressed as in Eq. (4-19) [44]:

$$x(n) = \sum_{k=0}^{M-2} \sum_{m=-\infty}^{\infty} y_k(m) \cdot g_k(n-2^{k+1} \cdot m) + \sum_{m=-\infty}^{\infty} y_{M-1}(m) \cdot g_{M-1}(n-2^{M-1} \cdot m) \quad (4.19)$$

The tree-structured dyadic synthesis filter bank, given in Fig. 4-27, can be employed to perform the inverse wavelet transform. Reconstruction low pass and high pass filters are cascaded to implement the Eq. 4.19. As it was in the case of analysis part there are only two different filters. The four filters in this structure are the basis of the wavelet transform. These analysis and synthesis filter bank structures can be extended to 8 channels.

The wavelet families in Matlab are tested in eight-band filter bank structure for hearing instruments. The wavelets simulated in this tree structure filter bank are: Haar,
Fig. 4-27 Dyadic synthesis filter bank structure

Daubechies, Symlets, Coiflets, Meyer, Discrete Meyer, Gaussian, Mexican hat, Morlet and Shannon. The channel gain tests are applied to the systems in Simulink environment with the configuration given in Fig. 4-28. The most reasonable magnitude response is obtained from Discrete Meyer wavelet. The magnitude responses of the channels are plotted together in Fig. 4-29.

Fig. 4-28 Dyadic filter bank structure in simulink for wavelet gain test
Fig. 4-29 Dyadic discrete meyer wavelet filter bank magnitude responses

A dyadic filter bank divides the spectrum into octave bands, which means it assigns half of the spectrum to the upper most channel and continues with the division of two of the rest. That means for this design the pass band frequencies are suppose to be: 8 kHz-4 kHz; 4 kHz-2 kHz ; 2 kHz-1 kHz ; 1 kHz-500 Hz; 500 Hz-250 Hz; 250 Hz-125 Hz; 125 Hz-72.5 Hz; 72.5 Hz-36.25 Hz. That means it is suppose to cover the spectrum of hearing up to 8 kHz. In Fig. 4-29 the gain in each channel is increased to 30 dB, while the rest is kept constant. We observe that the upper four channels satisfy the frequency expectations. The lower four channels disappear in the spectrum. This disappearance is because of the characteristic of the overall magnitude response plotted in Fig. 4-30. For low frequencies the response reaches to −40 dB. This characteristic is seen for all wavelet bases. The loss in low frequencies can be compensated but fluctuations in higher
frequencies will remain. Besides the problem in magnitude responses the phase response is extremely nonlinear and the group delay deviates from 500 samples to 2500 samples, which will delay the input up to 160 ms at 16 kHz sampling. Unfortunately this will have severe impact on sound.

Fig. 4-30 Overall Magnitude response of discrete meyer wavelet filter bank

4.5 DFT Modulated Filter Bank

In previous sections the spectrum of the sound is divided into pass bands through tree structured filter banks. In section 4.2 the data rate is kept as it is and in following sections the data rate is reduced to obtain a reduced cost of processing in channels. In this section a prototype filter is modulated with the aid of DFT and the signal is separated into bands with these modulated filters. The data rate is reduced with the aid of
analysis filter bank and the signal is recovered at its original rate through synthesis filter bank.

4.5.1  DFT Modulation Theory

The block diagram of a multirate filter bank can be given as in Fig. 4-31. In this block diagram H(z)'s are the analysis filters and G(z)'s are the synthesis filters. The analysis bank eliminates aliasing and the synthesis bank eliminates imaging, which are caused by the downsampling and upsampling by M. In this structure the filter designs can be independent to each other or they can be achieved from one prototype filter through complex modulation by multiplying the filter coefficients with $W_K^{-kn}$, which is defined in Eq. 4.20. ‘k’ is the band number and ‘n’ is the coefficient number, both starting from ‘0’.

$$W_K = e^{-j2\pi/k} \quad \text{and} \quad W_K^{-kn} = e^{j2\pi kn/k}$$  \hspace{1cm} (4.20)

‘K’ in Eq. 4.20 is the total number of channels in the filter bank structure. The effect of this modulation is demonstrated in Fig. 4-32. A low pass filter is modulated for K=16 and k=2.

Fig. 4-31 Multirate filter bank block diagram
It is observed that the filter's magnitude response is shifted and it becomes a bandpass filter, which can be used as an analysis filter for the third channel of an eight-band filter.

We achieve the same result if we modulate the input signal instead of the filter. That means the spectrum of the input signal is shifted, while the filter response is kept the same. From mathematical point of view the input signal is multiplied with the term in Eq. 4.20 no matter whether it is placed in filter coefficients or on signal path.

Since the modulation is complex we end up with complex samples after the analysis filter bank. In order to retrieve real samples, in synthesis filter bank demodulation should be applied. That means the prototype filter should be shifted to the opposite direction of analysis filters in frequency domain. The band pass filters are achieved by multiplying with $W_k^e$. 

![Complex modulation of a FIR filter](image)

Fig. 4-32 Complex modulation of a FIR filter

The block diagram of a modulated filter bank is given in Fig. 4-33. It is observed that the filters $h(n)$ and $f(n)$ are repeated in each channel [55]. The implementation of modulated
Fig. 4-33 Modulated filter bank structure

filter bank in this structure would be too inefficient. The same modulated filter bank can be implemented in polyphase structure.

**Polyphase Representation:** The polyphase representation of a transfer function is the separation of its coefficients in terms of their number. For example a filter with impulse response $h(n)$ can be defined as the addition of odd numbered and even numbered coefficients as follows \[44\]:

$$H(z) = \sum_{n=-\infty}^{\infty} h(2 \cdot n) \cdot z^{-2n} + z^{-1} \cdot \sum_{n=-\infty}^{\infty} h(2 \cdot n + 1) \cdot z^{-2n} \quad (4.21)$$

This separation can be performed for an integer K as follows:

$$H(z) = \sum_{l=0}^{K-1} z^{-l} \cdot E_l(z^K) \quad (4.22)$$

This representation has a special meaning because of the noble identities plotted in Fig. 4-34. The proof of these identities can be found in \[44\].
The analysis filter $h(n)$ and synthesis filter $f(n)$ in Fig.4-33 can be represented in polyphase form for $K=16$:

$$H(z) = \sum_{i=0}^{15} z^{-i} \cdot H_i(z^{16}) \quad F(z) = \sum_{i=0}^{15} z^{-i} \cdot F_i(z^{16})$$  \hspace{1cm} (4.23)

Using the noble identities in Fig. 4-34, the polyphase components can be replaced with downsamplers and upsamplers. The modulation can be taken after the analysis and before the synthesis filters. The final form of the DFT modulated filter bank is given in Fig. 4-35. A mathematical proof of the identity of the analysis part of Fig. 4-33 with the analysis part of Fig.4-35 can be done for critically sampled case ($M=K$) as follows:

$$p_k(n) = h(n) \cdot W^{-k \cdot n}_M, q_k(n) = f(n) \cdot W^{-k \cdot n}_M$$

$$X_k(m) = \sum_{n=0}^{L-1} h_k(n) \cdot x(m \cdot M - n) = \sum_{n=0}^{L-1} h(n) \cdot W^{-k \cdot n}_M \cdot x(m \cdot M - n)$$  \hspace{1cm} (4.24)

In Eq. 4.24 $h_k(n)$ are analysis filters and $q_k(n)$ are synthesis filters. $L$ is the length of the analysis filter. $X_k(m)$ are the channel signals. In final expression of $X_k(m)$ in Eq. 4-24 the substitutions $n = i \cdot M + j$ and $L = M \cdot L_p$ are made:
Fig. 4-35 DFT filter bank block diagram

\[ X_k(m) = \sum_{j=0}^{M-1} \sum_{i=0}^{L-1} h(i \cdot M + j) \cdot W^{-k(i \cdot M + i)} \cdot x(m \cdot M - i \cdot M - j) \]

\[ = \sum_{j=0}^{M-1} W^{-k,j} \sum_{i=0}^{L-1} h(i \cdot M + j) \cdot x(m \cdot M - i \cdot M - j) \]  

\[ = \sum_{j=0}^{M-1} W^{-k,j} \sum_{i=0}^{L-1} h_j(i) \cdot x_j(m - i) \]  

(4.25)

The final expression in Eq. 4.25 is IDFT of the convolution of the polyphase components of the analysis filter with polyphase components of input signal, which are basically delayed versions of the input signal [56]. It exactly describes the operation of the DFT filter bank in Fig. 4-34. The only difference from the normal IDFT is that the prefactor 1/M is omitted.

4.5.2 M-Band Filters

An M-Band noncausal filter is defined as having the following property [57]:

\[ \begin{align*}
    h(M \cdot p) &= 0 & p \neq 0 \\
    h(0) &= \frac{1}{M} & p = 0
\end{align*} \]

(4.26)
It is an extension of half band filters, which satisfy Eq. 3.3. The half band filters can be
designed successively by Remez algorithm but the design of M-band filters with this
algorithm doesn’t totally satisfy the conditions in 4.26. Therefore another design method
called ‘Eigenfilter’ method is used.

4.5.3 Eigenfilter Design Method

It is a least-squares FIR filter design method. Its difference from other methods is that
both frequency and time domain constraints can be put on filter design. For an even order
linear phase filter the frequency response can be written as:

\[ H_0(e^{j\omega}) = \sum_{n=0}^{M} b_n \cdot \cos(n \cdot \omega) \]  \hspace{1cm} (4.27)

Coefficients ‘b’ can be written in vector form:

\[ b = [b_0, b_1, b_M] \]  \hspace{1cm} (4.28)

An error matrix ‘P’ for the design of an even order filter can be constructed with a
weighting variable ‘\( \alpha \)’ for passband and stopband attenuation [58]:

\[
P(n,m) = \frac{(1-\alpha)}{\pi} \int_0^\pi (1-\cos(n \cdot \omega) \cdot (1-\cos(m \cdot \omega) \cdot d\omega
+ \frac{\alpha}{\pi} \int_0^\pi \cos(n \cdot \omega) \cdot \cos(m \cdot \omega) \cdot d\omega
\]  \hspace{1cm} (4.29)

The total error of the filter design is given by:

\[ E = b' \cdot P \cdot b \]  \hspace{1cm} (4.30)

The smallest eigenvalue of the matrix P is found and the eigenvector of this eigenvalue
gives the b vector, which has least squares minimum error for the design parameters.
The time constraints can be imposed to the method by keeping the rows and columns of the P matrix according to the desired value of the corresponding coefficient. That means to obtain a filter, whose 5th coefficient is zero the 5th column and row of the matrix is deleted. This allows us to design M-band filters. The design method is written in Matlab and the code is given in Appendix III. A more detailed explanation of the method can be found in [58].

4.5.4 Perfect reconstruction for DFT filter bank

The perfect reconstruction condition for critical subsampling (M=K) of the DFT filter bank is [56]:

\[ H_k(z) \cdot F_{M-1-k}(z) = \frac{z^{-N}}{M} \]  

(4.31)

In this equation N is an arbitrary integer. That means polyphase components of analysis filter bank and synthesis filter bank, whose delay sum gives M-1, are matched together and their cascade should give a delay. It is very difficult to design two filters forcing with these constraints.

M-band filters obtain the property:

\[ \sum_{k=0}^{M-1} H(z \cdot W^k) = 1 \]  

(4.32)

That means the summation of the modulated filters from an M-band filter will give unity. If the analysis and synthesis filters are designed as M-band filters and their convolution also gives an M-band filter, than perfect reconstruction is obtained in the filter bank [44].
This property is used in the design of the DFT filter bank, which is explained in the following section.

4.5.5 DFT Filter Bank simulation

An 8-band filter bank is designed employing two 16-band filters. The filter bank modulates the filters 16 times, but since only 8 of these bands lie in positive frequency region, the filter bank is called 8-band filter bank. The rest of the bands are necessary for reconstruction of the original signal. The simulator is written in Matlab's Simulink environment. The simulator is shown in Fig. 4-36.

Fig. 4-36 DFT Filter bank simulator in Simulink

The simulator has the same structure shown in Fig. 4-35. The analysis low pass filter is represented in polyphase form. The outputs of the polyphase elements are fed into IDFT block. Between IDFT and DFT gain is applied to the signal. This is the region where all
the signal processing should take place. After demodulation of the input signal with DFT
the signal is fed into polyphase components of the synthesis filter. The outputs are
delayed and summed up and the final output signal is obtained.

Using the eigenfilter design method a 16-band linear phase FIR filter is designed. The
magnitude response of this filter is plotted in Fig. 4-37. It is a 64th order filter.

![Magnitude response of the analysis filter](image)

**Fig. 4-37** Magnitude response of the analysis filter

This filter is used as analysis filter in the filter bank. It has a pass band cutoff frequency
of 240 Hz, pass band ripple of 0.1 dB, stopband cutoff frequency of 760 Hz. It doesn't
have an equiripple characteristic in the stopband. The stopband attenuation deviates from
40 dB to 57 dB.

Because of the high gain deviations in digital hearing aids, with critical sampling the
aliasing and imaging can not be eliminated totally. Therefore the signal is oversampled
by 2. That means instead of downsampling by 16, the signal is downsamped by 8 at analysis part and upsamped by 8 back to its original rate at synthesis part [59].

This relaxation allows us to design an 8-band filter in the synthesis part. This filter is extracted from the analysis filter by decimating the coefficients by 2. The magnitude response of the synthesis filter is plotted in Fig. 4-38. Because of the oversampling this frequency response is sharp enough to suppress the images of upsamping. The ripples of the filter are the same as analysis part. The passband cutoff frequencies are double of the analysis filter.

![Magnitude response of synthesis filter](image)

**Fig. 4-38** Magnitude response of synthesis filter

These filters have the property that their middle coefficient is an integer multiple of 16 away from the first coefficient. In Eq. 4.32 the real coefficient of W, which is ‘1’, should be multiplied with the middle coefficient of the M-band filter. Therefore the DFT and IDFT matrices should be shifted so that the polyphase element including the middle coefficient is fed to the path of the first row of W. Even though for this design the shift is
not necessary, the DFT and IDFT are directly implemented in the simulation. Instead of using the radix-2 algorithm the DFT coefficients are directly multiplied with the input signals.

The performance of the filter bank is tested using the gain increase method in pass bands. Since the filter bank is uniform and the characteristics of pass bands are exactly the same as the first band, the increase is applied only on the first channel of the structure. The gain is increased by 5 dB up to 40 dB and the response of the filter bank is plotted in Fig. 4-39.

![Graph](image)

Fig. 4-39 The effect of gain increase on DFT filter bank overall magnitude response

Almost the same response as in IIR filter bank is achieved. The effect on adjacent bands passes 3 dB after 30 dB gain. Therefore 30 dB gain should be the limit of the filter bank.

The overall magnitude response of the filter bank with no gain is plotted in Fig. 4-40. The ripple is in the range of -0.2 dB to +0.2 dB, which can be accepted as perfect reconstruction.
The total cost of the filter bank is determined by the cost of the analysis and synthesis filters and by the cost of the DFT and IDFT processes. They are tabulated below:

<table>
<thead>
<tr>
<th></th>
<th>Filters</th>
<th>IDFT+DFT</th>
<th>Total Cost</th>
</tr>
</thead>
<tbody>
<tr>
<td>Multiplication</td>
<td>92</td>
<td>128</td>
<td>220 (28 MPU)</td>
</tr>
<tr>
<td>Addition</td>
<td>62</td>
<td>256</td>
<td>318 (40 APU)</td>
</tr>
</tbody>
</table>

Table 4-3 DFT filter bank implementation cost

![Graph showing frequency vs. magnitude response](image)

Fig. 4-40 Overall magnitude response of the DFT filter bank

Both analysis and synthesis filters are linear phase as stated at their design process. Since in filter bank structure they are divided into their polyphase components, the linear phase property is partly lost. Fig. 4-41 shows the group delay of the filter bank at zero gain. It deviates between 45 samples and 48 samples. This much deviation is acceptable. When the gain is applied to the channels the deviation in group delay increases too. Fig. 4-42 shows the group delay of the system, when 30 dB gain is applied to the first channel.
Fig. 4-41 Group delay of the DFT filter bank with no gain

Fig. 4-42 Group delay of the DFT filter bank at 30 dB gain in first channel

There is a wider fluctuation of group delay between 36 samples and 53 samples. This will distort the sound signal.
4.6 Discussion

In this chapter research on digital filter banks is covered. Four different types of digital filter banks are designed and their performances are demonstrated.

The magnitude responses of extended QMF bank and wavelet filter bank don’t meet our frequency resolution criteria. In wavelet filter bank even the perfect reconstruction can not be achieved.

The IFIR and DFT modulated filter banks show similar magnitude responses. The cost of these two filter banks is almost the same in terms of MPU (Multiplication per unit time) and APU (Addition per unit time). IFIR has 27 MPU and 45 APU. On the other hand DFT filter bank has 28 MPU and 40 APU. Even though the MPU and APU numbers are close to each other, in DFT filter bank we need to have 220 multipliers and 318 adders in the hardware. Not all of them operate at the same time but their locations are independent to each other. Therefore IFIR filter bank takes less space than DFT filter bank.

Another important drawback of the DFT filter bank is the fact that it doesn’t have constant group delay. It causes distortion in the sound signal and the quality of the sound is degraded.

Because of the high gain need for the compensation of the moderate to severe hearing loss, 60 dB IFIR filter bank design is proposed as the optimum solution for the digital hearing instrument. Its processing delay is 6.4 ms at 16 kHz, which will allow 3.6 ms processing delay after the filter bank.

The quantization effects on 60 dB IFIR filter bank are examined. It is observed that 12-bit quantization causes increase in the noise floor. Therefore 16-bit quantization should be used.
Chapter 5

COMPRESSION ALGORITHMS AND PERFORMANCE EVALUATION

This chapter covers two state of the art digital compression algorithms, improvements achieved in them and the evaluation of their performances using a hearing loss simulator based on dynamic expansion. The compression algorithms are RangeEar and Homomorphic Multiplicative AGC (Automatic Gain Control). In order to make a healthy comparison the algorithms are simulated with the same filter bank. The choice of the previous chapter, which is the 8-band 60 dB IFIR filter bank, is employed in simulations.

5.1 RangeEar Compression Algorithm

The algorithm is separated into two parts. The first part is the frequency shaping of the input signal according to the audiogram. That means threshold level increase is compensated through amplification. The second part is the two-channel, wide dynamic range compression [60]. The block diagram of the system is given in Fig. 5-1.

5.1.1 RangeEar System Specifications

RangeEar employs 7-band IFIR filter bank with 40 dB stopband attenuation [61]. In filter bank design the second passband is divided into two passbands. On the other hand the two upper most passbands of the 8-band IFIR filter bank design in 4th chapter is left out
of the structure. At 17 kHz the width of the channels are listed in Fig. 5-1. The filter bank is designed in Matlab and the passbands are plotted in Fig. 5-2 at 17 kHz sampling frequency.

![Diagram](image)

**Fig. 5-1 RangeEar system block diagram**

![Graph](image)

**Fig. 5-2 Magnitude responses of RangeEar filter bank**
After the division of signals into bands, frequency shaping is applied in each channel. The three lower channels are summed up and fed into low channel compressor. The rest four bands are summed up and fed into high channel compressor. Before the compressors additional gain can be applied to the sums.

In compressor pure-tone loudness model is used [60]. The loudness model assumes a completion point. The loudness for the impaired ear and normal ear is the same above the completion point. Below the completion point, the loudness model is a linear approximation of the loudness growth curve given in Fig. 2-14. The completion point ‘M’ is defined in algorithm as:

\[ M = 0.8 \cdot HTL + 47dBSPL \]  \hspace{1cm} (5.1)

HTL is the hearing threshold level for the impaired person. The slope ‘S’ of the loudness growth function below M is determined as:

\[ S = (47 + 0.8 \cdot HTL)/(47 - 0.2 \cdot HTL) \]  \hspace{1cm} (5.2)

In low frequency compressor there are three HTL’s, which can be applied into algorithm. The algorithm takes the average of these three HTL levels as the HTL of the low compressor.

In high frequency compressor there are four HTL’s. The system feeds the lowest HTL into algorithm.

Both low frequency and high frequency compressors apply a knee point at 30 dBSPL, so that noise is not amplified. The operation of loudness growth function is demonstrated for three different HTL’s in Fig. 5-3. Normal sound level is mapped to impaired ears loudness perception. The compressors have 300 ms release time and 15 ms attack time.
5.1.2 RangeEar Simulator

The RangeEar digital hearing instrument described in section 5.1.1 is simulated in Matlab's Simulink environment. The simulator is shown in Fig. 5-4.
To achieve the 15 ms attack time and 300 ms release time DynamEQ-II’s twin average detection system is employed inside the compressors. The low frequency compressor block is shown in Fig. 5-5.

![Fig. 5-5 RangeEar low-frequency compressor block](image)

The slow and fast average detectors determine the intensity level of the input signal and they fed this value into RangeEar algorithm. The simulated I/O characteristics of the algorithm with increasing HTL are plotted in Fig. 5-6. The HTL is increased from 20 dBSPL by 10 dB up to 70 dBSPL. The completion point gets higher and the slope gets sharper with HTL increase. Input signals below 30 dBSPL are amplified linearly.

![Fig. 5-6 RangeEar compression algorithm I/O response with increasing HTL](image)
5.1.3 Modified RangeEar Simulator

In order to make a healthy comparison between the compression algorithms we have to use the same filter bank. Therefore the original RangeEar system is modified and the filter bank is replaced with the 60 dB IFIR filter bank design. The lower three bands are fed into low frequency compressor and the other five bands are processed by high frequency compressor. The compression algorithms are kept exactly the same.

5.2 Homomorphic Multiplicative AGC

The homomorphic multiplicative AGC (Automatic gain control) is based on a model of the response of the human auditory system to sound stimuli [62]. The model is developed from a similar model for the human visual system [63].

5.2.1 Theory of homomorphic multiplication

The block diagram of the loudness perception model is given in Fig. 5-7. According to the model the first operation is bandpass filtering by basilar membrane. Hair cells detect the intensity of the signal like an envelope detector in the band. After the envelope detection the intensity of the signal is logarithmically compressed. The high pass filter

![Diagram of the loudness perception model]

Fig. 5-7 Loudness perception model
provides loudness adaptation. The hyperbolic tangent mimics the firing of the inner hair cells. A multiplicative intrinsic noise source is added in the logarithmic domain. The exponential of the signal is taken and the loudness is detected by the neural network.

In this model the processing is done only on the envelope of the signal. That means if we define an acoustic signal \( s(t) \) as in Eq. 5.3, the \( v(t) \), which is the rapidly varying vibration is untouched [62].

\[
s(t) = e(t) \cdot v(t) \tag{5.3}
\]

The transfer function of the normal hearing is called as \( H_n(t) \) and the transfer function of the damaged ear is called as \( H_d(t) \). Their responses to an acoustic signal are defined as:

\[
n(t) = H_n(s(t)) \text{ and } d(t) = H_d(s(t)) \tag{5.4}
\]

In order to obtain \( n(t) \) in a damaged ear the response of the normal ear should be cascaded with the inverse response of the damaged ear:

\[
H_d(H_d^{-1}(n(t), v(t))) = n(t) \tag{5.5}
\]

In order to take the inverse response of the damaged ear, the information on fast varying part \( v(t) \) of the acoustic signal should be kept. Therefore \( v(t) \) is added to the transfer function. Eq. 5.5 shows that the audio processing we have to apply on input acoustic signal is given as:

\[
H_d^{-1}(n(t), v(t)) = H_d^{-1}(H_n(s(t)), v(t)) \tag{5.6}
\]

The inverse transfer function of damaged ear should be cascaded with the transfer function of the normal ear so that the desired response \( n(t) \) is obtained.
This cascade is expected to bring a very complex signal processing scheme, but most of the blocks in forward and inverse model cancel each other and the signal processing in one pass band is simplified to the form given in Fig. 5-8. The details of the simplification can be found in [62].

\[
\begin{align*}
H_k & \quad || \quad LPF \quad \frac{1}{e_{k,\text{max}}} \quad \log \quad K_k - 1 \quad \exp \quad \left( \frac{e_k(t)}{e_{k,\text{max}}} \right) \quad s_k(t) \\
\text{s}_k(t) &= \text{s}_k(t) \cdot v_k(t)
\end{align*}
\]

Fig. 5-8 The block diagram of homomorphic multiplicative AGC for one pass band

The system is called homomorphic multiplicative, because of the fact that after the intensity level is transformed into cepstrum domain by taking the natural logarithm, the signal is multiplied with \( K_k - 1 \). There isn’t any homomorphic filtering in the structure, but multiplication in cepstrum domain.

In Fig. 5-8, \( H_k \) is the band pass filter. The cascade of taking the absolute value of the signal and low pass filtering gives the envelope magnitude of the acoustic signal. The signal at the output of the envelope detector is normalized with \( e_{k,\text{max}} \), which is the UCL (upper comfortable level) of hearing. The normalized signal is mapped into cepstrum domain and homomorphically multiplied by \( K_k - 1 \). \( K_k \) is defined as:

115
\[ K_t = \left[ 1 - \left( \frac{\text{Hearing Loss (dBSPL)}}{\text{UCL (dBSPL)} - \text{NHT (dBSPL)}} \right) \right] \] (5.7)

The NHT is defined as 'Normal Hearing Threshold'. The processed signal is mapped back to time domain and multiplied with the original acoustical signal, which carries the fast varying signal (high frequency signal) information. The gain of the system is:

\[ \text{Gain}(t) = \left( \frac{e_k(t)}{e_k,\text{max}} \right)^{k_t^{-1}} \] (5.8)

The signal processing explained above is done in each channel of the digital hearing instrument separately.

The performance evaluation of this compression algorithm has been done with a 12-band filter bank at 21.33 kHz sampling frequency [64]. The subject tests showed better results comparing to other digital hearing instruments.

5.2.2 Homomorphic multiplicative AGC hearing instrument simulator

The compression algorithm is simulated in Matlab-Simulink. The 8-band 60 dB IFIR filter bank design is used for band pass filtering. The compression algorithm takes UCL, NHT and HL (Hearing Loss) as control parameters. The values of these parameters determine the I/O curve of the compressor. The simulator is shown in Fig. 5-9 and its code is given in Appendix IV. The compression is done in each channel independent to each other. Therefore there are 24 parameters to set in the hearing instrument. One of the compressor blocks is shown in Fig. 5-10. The low pass filter is a 10 ms long moving average detector. The compressor block is the exact implementation of Fig. 5-8. The \( e_{k,\text{max}} \) value is generated from the UCL level since it is the intensity value of UCL.
Fig. 5-9 Homomorphic multiplicative AGC Hearing Instrument Simulator

Fig. 5-10 Homomorphic multiplicative AGC compressor block

The effects of control parameters on compressors I/O curve are simulated. In first simulations the HL is varied, while the UCL is set at 110 dB SPL and NHT is set at 0 dB SPL. The effect of this sweep is plotted in Fig. 5-11.
Fig. 5-11 The effect of HL increase on homomorphic I/O curve

The HL is increased from 20 dBSPL to 50 dBSPL by 10 dB. The lower starting point of the I/O curve is shifted to the right, while the upper completion point is kept constant at 110 dBSPL. That means the compression ratio is increased to map the input dynamic range. The upper completion point is set as UCL.

In second part the effect of the change in UCL is simulated. The HL is kept constant at 30 dBSPL, NHT is kept constant at 0 dBSPL. The response is plotted in Fig. 5-12.

Fig. 5-12 The effect of UCL increase on homomorphic I/O curve
In Fig. 5-12 the UCL of the compressor is increased from 90 dBSPL to 110 dBSPL. The effect of this increase is the decrease in compression ratio. The completion point is increased, while the starting point remains at the same level.

The effect of the last control parameter NHT on compressor I/O curve is given in Fig. 5-13. This time UCL is kept constant at 110 dBSPL and HL is kept constant at 20 dBSPL.

![Graph showing effect of NHT increase on homomorphic I/O curve](image)

**Fig. 5-13 The effect of NHT increase on homomorphic I/O curve**

The NHT is increased by 10 dB from 0 dBSPL to 30 dBSPL. It is observed that this increase shifts the starting point of the I/O curve to right. The increase in the starting point is approximately 3 dB for every 10 dB increase in NHT.

### 5.2.3 Improved homomorphic multiplicative AGC algorithm

In previous section we observed that the only flexibility in I/O curve of the system is in starting point and completion point of the compression. No matter what the intensity level of the sound signal is, some degree of compression is applied on it. In hearing instruments the very low level signals shouldn’t face much amplification, so that the
noise at these levels are kept at their original level. Therefore there should be limits of compression.

The homomorphic multiplicative AGC applies constant gain in cepstrum domain. The signal is multiplied with $K_s - 1$ no matter what its intensity level is. The effect of taking the exponential after this multiplication causes the compression. At this cepstrum domain the gain can be varied according to the input intensity level too. A weighting factor is added to $K_s$. The factor increases with increasing intensity level of the input signal. The weighting factor is chosen as $IL/330$. $IL$ is the intensity level of the acoustic signal. That means $K_s$ is increased by an amount of 0 to 0.33, which depends on the input intensity level of the acoustic signal. The effect of this factor is plotted in Fig. 5-14.

![Graph showing improved homomorphic multiplicative AGC I/O curve](image)

**Fig. 5-14** Improved homomorphic multiplicative AGC I/O curve

In Fig. 5-14 the dashed line is the I/O curve of the original compression algorithm for HL = 30 dBSPL, UCL = 110 dBSPL and NHT = 0 dBSPL. The solid line is the response of new algorithm. To obtain this response the HL is pulled down to 10 dBSPL. We observe
that the improved response obtains almost the same characteristics for the input intensity levels higher than 60 dB SPL. For low level input range the signal is not overamplified. This is achieved by nonlinear compression rate.

The I/O curves in Fig. 5-14 are plotted again in terms of input and output voltages in Fig. 5-15. It is assumed that 110 dB SPL corresponds to 1 V. We observe that the two I/O characteristics converge to each other as the input level becomes 1V. Since we are using linear scale we can’t make a healthy observation on the responses below 70 dB SPL or 0.01 V. We know that the new algorithm applies more gain to signals over 70 dB SPL and this characteristic is also observed in Fig. 5-15.

![Graph showing comparison of input and output voltages](image)

**Fig. 5-15** Comparison of the homomorphic algorithms in terms of input, output voltages

The difference of these two I/O curves can be observed better on their compression ratio curves, which is given in Fig. 5-16. In Fig. 5-16 the dashed line shows the compression ratio of the original compression algorithm. We observe constant compression ratio over all the input dynamic range.
Fig. 5-16 Compression ratios of I/O curves of Fig. 5-14

The solid line is the compression ratio of the improved algorithm. It shows that the knee point for this response is 40 dBSPL. Over 40 dBSPL it starts the compression. Below this intensity level the system actually applies expansion to the system. Expansion is a common method used in communication channels to suppress the noise.

Another improvement in compressor system is obtained by implementing DynamEQ-II's twin average detection. With only one lowpass filter we have the same attack and release times. Twin average detection allows us to apply different attack and release times. The final form of the homomorphic multiplicative AGC compressor is given in Fig. 5-17.

Fig. 5-17 Improved homomorphic multiplicative AGC algorithm
On left hand side of the Fig. 5-17 the twin average detector is located. The output of this
detector is fed into compressor.

5.3 Hearing Loss Simulator

In order to evaluate the performances of the compression algorithms a digital hearing loss
simulator is employed. Input signal is presented to a normal listener at the same loudness
level, which would be perceived by an impaired listener. This is achieved by expansion
simulation. The I/O characteristic of the simulator is determined by three parameters. The
first parameter is $T_n$, which is the detection threshold of the normal hearing person. The
second parameter is the detection threshold of the impaired person $T_i$. The final
parameter is the threshold of recruitment $T_r$. This parameter is extracted from the first
two using the following formula [65]:

$$T_r = T_n + (T_i - T_n) \cdot \frac{\tan(\alpha)}{\tan(\alpha - 1)} \quad (5.9)$$

The angle of the recruitment curve $\alpha$ (in degrees) is defined as:

$$\alpha = 47 + 0.45 \cdot (T_i - T_n) \quad (5.10)$$

The effect of $T_n$ on expander I/O curve is plotted in Fig. 5-18. $T_i$ is kept constant at 30
dBSPL. That means the hearing loss is 30 dBSPL and the normal hearing threshold level
increases from 0 dBSPL to 25 dBSPL. In Fig 5-18 the increase of $T_n$ changes the lower
kneepoint of the expansion. Below this kneepoint linear gain is applied, above it the
signal is expanded. Besides the lower kneepoint, the change in $T_n$ also effects $T_r$, which
is the completion point of expansion.
The effect on $T_i$ is very slight and it is located approximately at 70 dBSPL. On the other hand the lower kneepoint is equal to the difference between the impaired hearing threshold and normal hearing threshold. It should be noted that the x-axis doesn't start from 0 dBSPL. That means the input signal is mapped to negative intensity levels too. The acoustic signal, which lies below the lower kneepoint, is mapped to a signal, which lies below the normal hearing threshold level. That means it won't be recognized by the listener. This is very logical, because the lower kneepoint is located at impaired hearing threshold level on input signal axis.

The effect of $T_i$, the impaired hearing threshold level, on expander I/O curve is demonstrated by increasing its value from 30 dBSPL to 55 dBSPL by 5 dB. The normal hearing threshold level $T_n$ is kept constant at 0 dBSPL. The I/O response is plotted in Fig. 5-19. The increase in $T_i$ shifts both the expansion completion point and the lower kneepoint up in y-axis, which is the input sound intensity level.
Fig. 5-19 $T_1$ parameter effect on loss simulator I/O curve

The knee point gets higher faster than the completion point, which brings higher expansion ratio. Above the completion point and below the knee point the gain is still constant.

The simulator has a 14-band filter bank. The filter bank consists of band pass filter with very high orders. The band pass filters are designed with Kaiser window method. They are 4000$^{th}$ order filters with very sharp transition bands and high stop band attenuations. The magnitude responses are plotted in Fig. 5-20. It divides the spectrum into octave bands. Because of their high order the magnitude responses look like boxcar windows.

The filter bank pass bands are plotted at 16 kHz sampling frequency in Fig. 5-20. The stop band attenuation is 75 dB. The noise floor doesn’t have a flat response. It decays very fast. The total magnitude response of the bank is maximally flat.

The hearing loss simulator employs 20 ms long windows to perform the level estimation of the acoustic signal. The moving averages are taken separately in each channel and the detected intensity levels are fed into gain blocks.
Fig. 5-20 Magnitude responses of loss simulator filter bank channels

Fig. 5-21 shows the implementation of hearing loss simulator in Matlab-Simulink. After the filter bank, the signals in each channel enter into level detectors. The output of this block, control parameters and the signal itself are fed into gain block, where the level dependant expansion is performed. The processed signals are summed up to construct the output of the loss simulator.

5.4 Performance Evaluation of Compression Algorithms

There isn’t any standard to evaluate the performance of hearing instruments. In our research performances of the digital hearing instruments are evaluated with simulations.

5.4.1 Simulation Set-up For Performance Evaluation

In first step an example audiogram is prepared for simulations. An acoustic signal is chosen and it’s fed into digital hearing instrument simulators, which are calibrated to compensate the hearing loss. The output of these hearing instruments is fed into digital
Fig. 5-21 Hearing Loss Simulator in Matlab-Simulink

hearing loss simulator, which maps the loudness level perception of an impaired ear to a normal ear. The block diagram of this set-up is given in Fig. 5-22.

Fig. 5-22 Performance evaluation set-up
Ideally the output of this evaluation should be equal to the input signal. The signal envelope should have similar amplitudes and vibrations as input signal envelope. If speech is taken as input acoustic signal, the output should be clearly understandable. On the other hand, in frequency domain the output signal magnitude should have similar characteristics of the input signal magnitude.

5.4.2 Simulations of performance evaluation

For the first simulation the impaired hearing assumed to have the audiogram in Fig. 5-23.

![Graph of Hearing Loss vs Frequency](image)

*Fig. 5-23 First hearing loss audiogram for performance evaluation*

The audiogram frequencies and hearing losses are given in Table 5-1. This has the characteristics of the noise induced hearing loss, which is pointed out in section 2.3.2.1. There is a sharp decrease in hearing around 2 kHz.

In first simulation a speech signal is chosen as the acoustic signal. The time waveform and spectrum of the speech signal are given in Fig. 5-24.
<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1000</th>
<th>2000</th>
<th>4000</th>
<th>8000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hearing Loss (dB)</td>
<td>10</td>
<td>10</td>
<td>15</td>
<td>10</td>
<td>40</td>
<td>30</td>
<td>20</td>
</tr>
</tbody>
</table>

Table 5-1 Audiogram values for first part of simulations

Fig. 5-24 Time waveform and spectrum of the speech signal for performance evaluation

It is shown in chapter 3 that most of the frequency content of the speech signal lies below 3kHz. The same characteristic is observed for this example too.

To compare the performances of the compression algorithms we have to take the simulation results both in frequency domain and in time domain into consideration. It is very difficult to observe the differences between the responses, if we compare them on the graphs in Fig. 5-24.

It is pointed out in section 5.2.1 that the information bearing part of the input speech is its envelope. Therefore the envelope of the signal is extracted by taking the absolute value of the signal and applying a 30ms long moving average detector on this rectified signal. The same technique is applied also on the spectrum of the signal. The envelope of the time
The waveform of the speech signal is given in Fig. 5-25. The envelope of the spectrum of this acoustic signal is given in Fig. 5-26.

![Speech waveform envelope extracted as average amplitude](image1)

Fig. 5-25 Speech waveform envelope extracted as average amplitude

![Speech spectrum envelope extracted as average magnitude](image2)

Fig. 5-26 Speech spectrum envelope extracted as average magnitude

The first simulation is performed with RangeEar algorithm. The frequency shaping parameters are adjusted according to the sample audiogram given in Fig. 5.23. As shown in evaluation set-up in Fig. 5-22 the output of the hearing instrument is fed into hearing loss simulator, whose impaired hearing threshold parameters are set according to the
audiogram. Normal hearing threshold is taken as 0 dB SPL. The output of evaluation structure is 2103 samples delayed version of the input signal. It is because of the group delay of the hearing instrument filter bank (103 samples) and hearing loss simulator filter bank (2000 samples). To make a healthy comparison the first 2103 samples of the output signal are ignored and the output is plotted starting the 2104th sample. The time waveform envelope of the evaluation for RangeEar algorithm is plotted in Fig. 5-27. The solid line shows the output of the system with RangeEar algorithm, whereas the dashed line is the speech envelope given in Fig. 5-25. The amplitudes are not converted to corresponding intensity levels, so that the small variations can be observed clearly.

![Figure 5-27 RangeEar time waveform envelope for speech signal](image)

In Fig. 5-27 we observe that the RangeEar envelope deviates around the desired envelope. It gets higher than the desired response around 0.7, 1.7 and 3 sec. On the other hand it lies below the desired envelope most of the time. Because of these fluctuations the
shape of the envelope is distorted. This effects the quality of the speech. In terms of
intelligibility it is preferred to have a response slightly higher than the desired response.
The RangeEar response drops down to almost half of the target response at 1 sec.

The RangeEar spectrum envelope is given in Fig. 5-28. The dashed line shows the target
envelope, which is plotted in Fig. 5-26.

![Fig. 5-28 RangeEar spectrum envelope for speech signal](image)

For low frequencies the approximately 10 dB hearing loss can not be compensated
successfully with RangeEar algorithm. On the other hand excessive gain is applied to the
region between 1 kHz to 2 kHz, where the sharp decrease of hearing threshold occurs.
After 3 kHz most of the upper midrange components, which are improving the
intelligibility of speech, are lost.
The same speech signal is fed into test system with the homomorphic multiplicative AGC algorithm. The settings of the hearing loss simulator are kept the same and the parameters of the algorithm are set to compensate the hearing loss given in Fig. 5-23.

The time waveform envelope of the test system output with response to the input speech signal is plotted in Fig. 5-29. The same method for RangeEar evaluation is used as the desired response is plotted with dashed lines.

![Homomorphic multiplicative AGC time waveform envelope for speech signal](image)

Fig. 5-29 Homomorphic multiplicative AGC time waveform envelope for speech signal

We observe that the envelope is constantly higher than the desired envelope. It has almost the same shape as the original speech envelope. The important speech clues are successfully kept. The only defect is the slightly higher intensity level at the output. This defect can be useful in an environment where the speech is getting masked.
The homomorphic multiplicative AGC spectrum envelope is given in Fig. 5-30. The dashed line shows the desired signal average magnitude.

![Graph showing spectrum envelope](image)

**Fig. 5-30 Homomorphic multiplicative AGC spectrum envelope for speech signal**

It is observed that the reason of the higher intensity levels in Fig. 5-29 is the excessive amplification of low frequency components up to 2 kHz in Fig. 5-30. The hearing loss after 2 kHz is recovered to the normal level. At higher frequencies above 6 kHz the signal faces an excessive amplification too. The processed envelope has similar shape to the original one.

For the second set of simulations another audiogram is chosen. The audiogram is given in Fig. 5-31. It shows moderate hearing loss at low frequencies up to 1 kHz. Between 1 kHz and 2.5 kHz the loss is slight. It gets again to moderate level after 2.5 kHz.

The performances of algorithms are examined with another most commonly heard acoustic signal, the music.
Fig. 5-31 Second hearing loss audiogram for performance evaluation

The audiogram values for the second set of simulations are listed in table 5-2:

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1000</th>
<th>2000</th>
<th>4000</th>
<th>8000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hearing Loss (dB)</td>
<td>50</td>
<td>45</td>
<td>40</td>
<td>20</td>
<td>10</td>
<td>50</td>
<td>60</td>
</tr>
</tbody>
</table>

Table 5-2 Audiogram values for second part of evaluation

The time waveform and spectrum of the music signal are plotted in Fig. 5-32.

Fig. 5-32 Time waveform and spectrum of the music signal for performance evaluation
From the time waveform of the signal we observe that there is a sudden increase in the intensity level of the signal around 2.2 sec. This is one of the most difficult signals to handle with for hearing instruments. The reason is explained in section 3.2.1. In spectrum of the signal low frequency components are dominant.

The envelopes in time domain and frequency domain are extracted using the same length windows as in first set of simulations. The envelope of the time waveform is given in Fig. 5-33. The spectrum envelope is shown in Fig. 5-34.
The music signal is applied to the test system given in Fig. 5.22, which employs the RangeEar algorithm for the first part. The envelope of the output signal is plotted together with the envelope of the input acoustic signal in Fig. 5-35. The solid line gives the RangeEar test system output.

![RangeEar time waveform envelope for music](image)

**Fig. 5-35 RangeEar time waveform envelope for music**

In Fig. 5-35 we observe that the RangeEar envelope is slightly over the desired envelope up to 2.2 sec. After this point the input signal gets a sharp increase and in response to this increase RangeEar envelope also gets a higher value. This time the difference between the desired envelope and RangeEar envelope gets really high. Since twin average detection system recovers really quickly to the transition, this difference is not because of the attack time. The I/O curve of RangeEar in Fig. 5-6 shows that there isn't any limiting applied to high level signals. That means the high intensity signals are tried to be kept as they are, but when a high level signal is multiplied with a slight different factor it can cause discomfort.
The spectrum envelopes of the same simulation are given in Fig. 5-36.

![Graph showing spectrum envelopes](image)

Fig. 5-36 RangeEar spectrum envelope for music signal

The reason of the big difference between the two envelopes in Fig. 5-35 is explained by Fig. 5-36. The RangeEar algorithm applies too much gain to the frequency ranges, which suffer from moderate or severe hearing loss. In the simulation with speech signal the moderate hearing loss is between 1 kHz and 3 kHz and RangeEar spectrum envelope is suffering in that range as shown in Fig. 5-28. For this simulation the same frequency range is the only part of the spectrum with slight hearing loss. Therefore the RangeEar envelope lies under the desired envelope only in this region. In the rest of the spectrum it is above the desired response.

The response of the system with homomorphic multiplicative AGC algorithm to music signal is observed. The first plotting is the time waveform envelopes of homomorphic signal and desired signal. They are plotted in Fig. 5-37.
The homomorphic multiplicative AGC algorithm shows the same characteristics it has shown for speech signal. The signal envelope characteristics are preserved and the algorithm response is slightly higher than the desired response. The effect of the algorithm is also examined in frequency domain as plotted in Fig. 5-38.
In Fig. 5-38 we observe that the homomorphic spectrum envelope almost coincides with the desired envelope. For the slight hearing loss range the spectrum envelope is slightly lying below the desired envelope. At the upper frequency moderate hearing loss range the envelope of homomorphic algorithm signal is slightly above the desired spectrum envelope.

5.5 Discussion

The RangeEar algorithm has the comfort of setting only the hearing loss amount read from the audiogram into frequency shaping. The algorithm makes the necessary compression adjustments by itself. The performance evaluation simulations show that the calibration is not very successful. There are couple reasons for that. The most important one is the fact that the algorithm applies two-channel compression. For a hearing loss with sharp slopes the compression should be different at very close frequencies. That means not only for frequency shaping, also for compression we need high frequency resolution. The reason of using two-channel compression system is not to distort the input speech signal. It has been seen that for hearing losses with rapidly varying audiograms the two-channel compression is more harmful than protective for speech. Taking the averages of three channels or the maximum of four channels into consideration degrades the frequency resolution for compression drastically.

The original homomorphic multiplicative AGC suffers from the fact that it doesn’t utilize any kneepoints of compression. From the starting point till completion point the compression ratio is constant. This problem is solved by introducing nonlinear compression ratio through the addition of weighting function in cepstrum domain. The
signals below the lower knee point are expanded. On the other hand, when the I/O curve gets closer to the completion point of the algorithm, which is the UCL, the compression ratio increases rapidly and applies compressive clipping to the input signal. That means by using high compression ratios the output signal is not allowed to pass the completion point. The performance of the algorithm is observed in simulations. The response is slightly higher than the desired one, but carries the same envelope characteristics.

In simulations both RangeEar and homomorphic multiplicative AGC algorithms are equipped with twin average detection system. The successful operation of this detection system is observed especially in music signal simulations.

The quantitative observations show the improved homomorphic multiplicative AGC algorithm as the best choice. The qualitative analysis, which is performed by listening the output of the hearing loss simulator, supports the choice.

5.6 Overall Discussion

DynamEQ-II hearing instrument has the drawback of low frequency resolution because of its two-band filter bank structure. Therefore digital filter bank algorithms are studied and 8-band 60 dB IFIR (interpolated finite impulse response) filter bank is selected as the optimum solution. The MPU (multiplications per unit time), APU (additions per unit time) and group delay characteristics of the studied filter bank structures determine the choice. Two state-of-the-art compression algorithms are studied and a new improved compression algorithm is achieved by intensity level dependant homomorphic multiplication. The improvement is demonstrated by employing MIT hearing loss simulator. Both quantitative and qualitative analyses support the choice of modified
homomorphic multiplicative AGC (automatic gain control). It is observed that multichannel signal processing is necessary not only for frequency shaping but also dynamic range compression.
Chapter 6

CONCLUSIONS

1. A sophisticated SIMULINK model, involving the use of audio files (16 bit *.wav), was developed to evaluate the performance characteristics of the strategies and algorithms used in the DynamEQ-II hearing instrument from the Gennum Corporation of Burlington, Ontario. The simulator exhibited the same input-output audio responses as the DynamEQ-II.

2. The two-band filter bank used in the DynamEQ-II hearing instrument limits the resolution in the frequency domain required to compensate hearing loss more accurately. A high resolution digital filter bank has been developed that uses eight distinct bands designed using an interpolated finite impulse response (IFIR) prototype filter that has been optimized using delay elements to reduce the phase distortion so as to give a maximally flat overall magnitude response. The resulting group delay is a constant and less than the value where self-hearing and "lip reading" problems would occur.

3. A sophisticated SIMULINK model, involving the use of audio files (16 bit *.wav), was developed to evaluate the performance characteristics of the RangeEAr algorithm as employed in the DigiFocus hearing instrument from the Oticon Company. The simulator incorporated the compression algorithm based on the pure tone loudness model.
4. An existing homomorphic multiplicative compression algorithm to provide automatic gain control (AGC) was studied and improved. The new compression algorithm is also based on a model of the human auditory system, but instead of multiplying the signal, after it has been transformed into the Cepstrum Domain, by a constant the new algorithm utilizes an acoustic signal intensity level weighted multiplication. The resulting nonlinear compression ratio expands low level signals and compresses high level signals in such a manner as to improve the signal to noise ratio and the intelligibility of the sound.

5. The use of the MIT hearing loss simulator was found to be a comprehensive and sophisticated tool to evaluate the effectiveness of the new proposed digital filter bank and compression algorithm for use in a hearing instrument, through both analysis and listening to actual test audio files.

6-1 Future Work

The cascade of the 8-band filterbank with 60 dB noise floor and modified homomorphic multiplicative AGC should be implemented in hardware to evaluate the performance with field test, where patients with sensorineural hearing loss are employed.

The hardware implementation of these algorithms can employ MEMS microphones and actuators to carry the hearing instrument further inside the ear into middle ear, as explained in [66].
The control of the parameters of the digital hearing instrument can be achieved through wireless technology, which can co-operate with internet to bring more comfort to the hearing impaired individual.
REFERENCES


[65] David S. Lum, Louis D. Braida, “DSP Implementation of a Real-Time Hearing Loss Simulator Based on Dynamic Expansion”, Research Laboratory of Electronics, Massachusetts Institute of Technology.

Appendix I

MATLAB CODE FOR HRTF DATA PLOTTING

% Using Shaw's azimuthal dependance results magnitude responses
% of different azimuths are plotted
T1 = [ 
0 0 0 0 0 0 0 0 0 0 0 0 0 0; 
0.4 0.4 0.5 0.5 0.8 1.1 1.4 1.4 1.4 1.4 1.4 1.3 1.4 1.5 1.5 1.4; 
0.7 0.8 1.0 1.1 1.7 2.2 2.6 2.6 2.6 2.6 2.6 2.6 2.7 2.7 2.8 2.7; 
1.0 1.2 1.4 1.5 2.4 3.2 3.6 3.6 3.5 3.5 3.5 3.7 3.8 4.0 3.9 3.7; 
1.3 1.5 1.8 1.9 2.8 3.8 4.3 4.3 4.3 4.4 4.5 4.7 4.9 4.9 4.9 4.7; 
1.5 1.7 2.1 2.2 3.2 4.3 4.8 4.8 4.8 4.8 4.9 5.1 5.3 5.7 5.7 5.5; 
1.6 1.8 2.2 2.3 3.4 4.5 5.0 5.0 5.0 5.1 5.3 5.6 6.1 6.1 5.7; 
1.6 1.8 2.1 2.2 3.3 4.3 4.8 4.9 4.9 5.0 5.2 5.5 6.1 6.0 5.5; 
1.4 1.6 1.9 2.0 3.0 3.9 4.4 4.5 4.6 4.7 5.0 5.4 6.0 5.9 5.4; 
1.1 1.3 1.5 1.6 2.4 3.1 3.5 3.6 3.8 4.1 4.5 5.0 5.9 5.9 5.4; 
0.7 0.8 0.9 0.9 1.4 2.0 2.4 2.5 2.7 3.2 3.7 4.3 5.2 5.2 5.0; 
0.2 0.1 0 0 0.3 0.7 1.1 1.2 1.4 1.9 2.6 3.3 4.1 4.1 3.9; 
-0.2 -0.5 -0.7 -0.7 -0.6 -0.4 -0.3 -0.1 0.4 1.1 1.8 2.3 2.3 2.1; 
-0.5 -0.9 -1.2 -1.3 -1.6 -1.7 -1.6 -1.6 -1.4 -1.2 -0.8 -0.3 0.2 0.2; 
-0.8 -1.3 -1.7 -1.8 -2.2 -2.5 -2.6 -2.6 -2.7 -2.8 -2.9 -2.8 -2.3 -2.2; 
-2.0; 
-1.1 -1.5 -1.8 -1.9 -2.2 -2.5 -3.0 -3.1 -3.5 -3.9 -4.3 -4.5 -4.5 -4.5; 
-1.4 -1.6 -1.7 -1.7 -1.9 -2.0 -2.5 -2.6 -2.9 -3.2 -3.4 -3.6 -3.9 -4.1; 
-1.6 -1.6 -1.5 -1.5 -1.4 -1.4 -1.7 -1.8 -1.9 -2.0 -2.0 -2.0 -2.1 -2.4; 
-1.6 -1.5 -1.4 -1.4 -1.2 -1.0 -1.2 -1.3 -1.5 -1.5 -1.4 -1.2 -1.0 -1.1; 
-1.5 -1.4 -1.4 -1.4 -1.4 -1.4 -1.6 -1.7 -2.0 -2.3 -2.4 -2.5 -2.9 -3.1; 
-4.0; 
-1.3 -1.3 -1.4 -1.4 -1.8 -2.2 -2.5 -2.6 -2.9 -3.4 -3.9 -4.4 -5.9 -6.3; 
-7.1; 
-1.0 -1.1 -1.3 -1.4 -1.9 -2.4 -2.9 -3.0 -3.4 -4.1 -4.8 -5.4 -6.3 -6.3; 
-6.1; 
-0.7 -0.8 -1.0 -1.1 -1.5 -2.0 -2.4 -2.5 -2.8 -3.2 -3.6 -3.9 -4.0 -3.9; 
-3.6; 
-0.4 -0.4 -0.5 -0.5 -0.8 -1.1 -1.3 -1.3 -1.4 -1.6 -1.8 -1.9 -1.8 -1.8; 
-1.6; ];

T2 = [ 
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0; 
1.3 1.2 1.2 1.3 1.5 1.8 2.0 2.1 2.2 2.2 2.1 2.1 2.4 2.6 2.6; 
2.5 2.3 2.3 2.4 2.4 3.1 3.5 3.7 3.9 3.7 3.5 3.8 4.5 5.0 5.1; 
3.4 3.3 3.0 3.2 3.6 3.6 3.6 4.1 3.6 4.3 4.2 4.6 4.6 5.5 6.8 7.4; 
4.2 3.7 3.3 3.5 3.5 4.0 4.6 4.9 4.8 4.6 3.8 3.3 3.4 4.4 7.7 8.8; 
4.6 3.6 2.9 3.0 3.5 3.6 4.1 4.0 3.5 2.5 1.8 3.0 5.6 7.9 9.4; 
4.5 3.1 2.0 1.7 2.1 2.4 2.4 2.3 1.9 0.7 -0.3 0.8 4.0 7.0 9.3; ]
4.1 2.2 0.8 -0.2 -0.2 0 0.1 0.1 -0.1 -1.2 -2.4 -1.7 1.7 4.8 7.7 ;
3.9 2.0 0.2 -1.3 -1.5 -1.4 -1.2 -1.2 -1.4 -2.3 -4.3 -4.2 -1.0 2.0 4.8 ;
4.0 2.2 0.4 -1.6 -1.9 -1.8 -1.8 -1.9 -2.7 -5.2 -6.0 -3.8 -0.8 1.7 ;
3.8 2.2 0.6 -0.9 -1.5 -1.4 -1.3 -1.3 -1.4 -2.5 -4.9 -5.9 -4.9 -3.0
-1.0 ;
2.8 1.2 -0.2 -1.2 -1.5 -1.6 -1.5 -1.4 -1.5 -2.4 -4.3 -5.3 -4.9 -4.2
-3.3 ;
1.3 -0.2 -1.6 -2.5 -2.6 -2.5 -2.5 -2.6 -3.4 -5.0 -5.4 -4.5 -5.0
-5.1 ;
-0.4 -2.0 -3.4 -4.3 -4.4 -4.3 -4.2 -4.2 -4.4 -5.3 -6.7 -6.6 -5.8 -6.4
-6.7 ;
-2.4 -3.8 -5.4 -6.5 -6.6 -6.6 -6.7 -6.7 -6.9 -7.8 -9.1 -9.2 -8.7 -9.0
-9.4 ;
-4.9 -6.2 -7.8 -9.0 -9.1 -9.1 -9.1 -9.2 -9.5 -10.6 -12.3 -12.2 -11.4
-11.6 -11.9 ;
-7.1 -8.9 -10.0 -10.5 -10.6 -10.6 -10.8 -10.9 -11.3 -12.5 -13.8 -13.6
-12.8 -12.3 -12.5 ;
-5.8 -7.7 -9.0 -9.7 -10.0 -10.2 -10.6 -10.8 -11.3 -12.5 -13.8 -13.6
-12.6 -12.0 -12.2 ;
-11.5 -11.4 ;
-5.9 -6.8 -7.0 -6.8 -7.0 -7.2 -7.7 -8.1 -8.9 -10.7 -12.9 -12.7 -11.7
-11.3 -11.1 ;
-7.3 -6.6 -6.0 -6.2 -6.6 -6.9 -7.0 -7.2 -8.0 -10.0 -11.9 -11.9 -11.1
-11.0 -10.9 ;
-5.3 -4.5 -4.3 -5.1 -5.8 -6.0 -6.0 -6.1 -6.7 -8.1 -9.3 -9.0 -8.6 -8.8
-6.9 ;
-3.1 -2.8 -2.7 -3.4 -3.9 -4.2 -4.4 -4.5 -4.8 -5.2 -5.8 -5.7 -5.6 -5.6
-5.5 ;
-1.4 -1.3 -1.3 -1.6 -1.9 -2.1 -2.3 -2.4 -2.4 -2.5 -2.6 -2.6 -2.6
-2.6 ;
T3 = [
0 0 0 0 0 0 0 0 0 0 0 0;
2.6 2.6 2.4 2.3 2.1 2.1 2.1 2.2 2.3 2.4 2.3 2.0 1.2 ;
5.1 5.0 4.8 4.4 4.0 3.8 3.7 3.7 3.7 3.6 3.3 2.8 1.6 ;
7.4 7.3 7.0 6.3 5.6 5.1 4.7 4.4 4.4 4.3 3.7 3.0 1.6 ;
8.9 8.9 8.5 7.9 7.0 6.3 5.6 5.0 4.7 4.4 4.0 3.3 2.3 ;
9.9 10.0 9.7 9.1 8.1 7.2 6.4 5.9 5.4 5.2 5.1 5.0 5.0 ;
10.1 10.3 10.0 9.5 8.7 7.7 7.0 6.4 6.2 6.4 6.2 6.2 6.2 6.2 6.2 6.2 6.2 6.2 ;
8.8 9.2 9.0 8.6 7.9 7.2 6.7 6.4 6.4 6.6 6.7 6.3 8.0 9.0 ;
6.3 7.0 7.2 6.9 6.3 5.9 5.5 5.5 5.5 5.7 6.0 6.7 7.3 8.0 ;
3.3 4.0 4.6 4.3 3.8 3.4 3.3 3.6 4.0 4.5 5.1 5.6 6.2 ;
0.1 0.7 1.8 1.6 1.0 0.7 0.8 1.1 1.6 2.1 2.6 3.1 3.7 ;
-2.7 -2.2 -0.8 -1.0 -1.4 -1.8 -1.8 -1.5 -1.0 -0.7 -0.2 -0.2 0.5 ;
-4.8 -4.5 -3.2 -3.2 -3.8 -4.2 -4.4 -4.4 -4.0 -3.7 -3.3 -2.9 -2.5 ;
-6.6 -6.4 -5.7 -5.6 -5.9 -6.4 -6.4 -6.4 -6.0 -6.2 -6.1 -4.9 -4.8 ;
-11.7 -11.4 -10.6 -10.2 -10.5 -11.0 -11.9 -12.5 -12.9 -13.0 -12.9
-12.4 -11.6 ;
-12.4 -12.1 -11.5 -11.2 -11.5 -12.3 -13.3 -14.1 -14.6 -14.8 -14.8
-14.3 -13.9 ;
-12.0 -11.7 -10.7 -10.4 -10.9 -11.9 -13.0 -13.9 -14.5 -14.8 -15.0
-14.8 -14.5 ;
-14.2 ;
-10.8 -10.4 -8.8 -8.0 -8.0 -8.9 -10.0 -11.0 -11.8 -12.1 -12.3 -12.2
-12.0 ;
-10.6 -10.2 -8.8 -8.2 -8.3 -9.0 -10.0 -11.0 -11.3 -11.3 -11.0 -10.0
-9.0 ;
-8.7 -8.4 -7.8 -7.6 -7.7 -8.7 -9.8 -10.7 -10.8 -10.4 -9.6 -8.1 -6.5 ;
-5.5 -5.5 -5.4 -5.2 -5.0 -5.4 -6.3 -7.0 -7.3 -7.0 -6.4 -5.1 -4.0 ;
-2.6 -2.6 -2.6 -2.5 -2.4 -2.4 -2.7 -3.0 -3.1 -3.0 -2.8 -2.3 -1.6 ;
]

F1 = [ 0.5 1.0 1.3 1.4 1.5 1.8 2.3 2.4 2.8 3.1 3.0 2.6 2.7 3.0 4.1 ];

F2 = [ 6.1 9.0 12.0 15.9 16.8 16.8 15.8 15.4 14.9 14.7 14.3 12.8 10.7 8.9 7.3 ];

F3 = [ 6.4 5.8 4.3 3.1 1.8 0.5 -0.6 -1.7 -1.7 2.5 6.8 8.4 8.5 ];

F = [ F1 F2 F3 ];

a = [0:15:345];

D = [T1 T2 T3];

FF = [FF1 FF2 FF3];

Fav=0;

for m=1:13
    Fav=Fav+D(m,:);
end

Fav=Fav./13;

figure

plot(F,Fav)

title('Average Frequency Response in the range of 0 to 180 degree');
ylabel('Magnitude Response (dBSPL)');
xlabel('Frequency(kHz)');

F30=D(3,:)+FF;
F60=D(5,:)+FF;
F90=D(7,:)+FF;
F120=D(9,:)+FF;
F150=D(11,:)+FF;
F180=D(13,:)+FF;

figure

plot(F,FF,'y',F,F30,'m',F,F60,'c',F,F90,'r',F,F120,'g',F,F150,'b',F,F180,'k')
Appendix II

OPTIMIZATION OF DELAY CHARACTERISTICS IN IFIR

load h1
load h1c
load h2
load h2c
load h3
load h3c
load h4
load h4c
load h5
load h5c
load h6
load h6c
load h7
load h7c
load h8
B1=conv(conv(h1,h2),conv(h4,h8));
B2=conv(conv(h1,h2),h4c);
B3=conv(conv(h1,h2c),h5);
B4=conv(conv(h1,h2c),h5c);
B5=conv(conv(h1c,h3),h6);
B6=conv(conv(h1c,h3),h6c);
B7=conv(conv(h1c,h3c),h7);
B8=conv(conv(h1c,h3c),h7c);
[H1,F] = FREQZ(B1,1,2000,12000);
[H3,F] = FREQZ(B3,1,2000,12000);
[H4,F] = FREQZ(B4,1,2000,12000);
[H5,F] = FREQZ(B5,1,2000,12000);
[H6,F] = FREQZ(B6,1,2000,12000);
[H7,F] = FREQZ(B7,1,2000,12000);
[H8,F] = FREQZ(B8,1,2000,12000);
Ec=100;
for i=0:24
    d=zeros(1,1+i);
    d(1+i)=1.0;
    B2d=conv(B2,d);
    [H2,F] = FREQZ(B2d,1,2000,12000);
    H=H1+H2+H3+H4+H5+H6+H7+H8;
    I=ones(250,1);
    E=sum(abs(I-abs(H(876:1125))));
    if E<Ec
        Ec=E;
        In=i;
    end
end
Ec=100;
In

155
d2=zeros(1,12);
d2(12)=1;
B2d=conv(B2,d2);
[H2,F] = FREQZ(B2d,1,2000,12000);
for i=0:24
    d=zeros(1,1+i);
    d(1+i)=1.0;
    B3d=conv(B3,d);
    [H3,F] = FREQZ(B3d,1,2000,12000);
    H=H1+H2+H3+H4+H5+H6+H7+H8;
    I=ones(675,1);
    E=sum(abs(I-abs(H(451:1125))));
    if E<Ec
        Ec=E;
        In2=i;
    end
end
In2
Ec=100;
d2=zeros(1,12);
d3=zeros(1,13);
d5=zeros(1,13);
d2(12)=1;
d3(13)=1;
d5(13)=1;
B2d=conv(B2,d2);
B3d=conv(B3,d3);
B5d=conv(B5,d5);
[H2,F] = FREQZ(B2d,1,2000,12000);
[H3,F] = FREQZ(B3d,1,2000,12000);
[H5,F] = FREQZ(B5d,1,2000,12000);
for i=0:24
    d=zeros(1,1+i);
    d(1+i)=1.0;
    B1d=conv(B1,d);
    [H1,F] = FREQZ(B1d,1,2000,12000);
    H=H1+H2+H3+H4+H5+H6+H7+H8;
    I=ones(1125,1);
    E=sum(abs(I-abs(H(1:1125))));
    if E<Ec
        Ec=E;
        In4=i;
    end
end
In4
Ec=100;
d2=zeros(1,12);
d3=zeros(1,13);
d5=zeros(1,13);
d1=zeros(1,11);
d2(12)=1;
d3(13)=1;
d5(13)=1;
d1(11)=1;
B2d=conv(B2,d2);
B3d=conv(B3,d3);
B5d=conv(B5,d5);
B1d=conv(B1,d1);
[H2,F] = FREQZ(B2d,1,2000,12000);
[H3,F] = FREQZ(B3d,1,2000,12000);
[H5,F] = FREQZ(B5d,1,2000,12000);
[H1,F] = FREQZ(B1d,1,2000,12000);
for i=0:24
  d=zeros(1,1+i);
  d(1+i)=1.0;
  B4d=conv(B4,d);
  [H4,F] = FREQZ(B4d,1,2000,12000);
  H=H1+H2+H3+H4+H5+H6+H7+H8;
  I=ones(1550,1);
  E=sum(abs(I-abs(H(1:1550))))
  if E<Ec
    Ec=E;
    In5=i;
  end
end
In5

In5

d2=zeros(1,12);
d3=zeros(1,13);
d5=zeros(1,13);
d1=zeros(1,11);
d4=zeros(1,13);
d2(12)=1;
d3(13)=1;
d5(13)=1;
d1(11)=1;
d4(13)=1;
B2d=conv(B2,d2);
B3d=conv(B3,d3);
B5d=conv(B5,d5);
B1d=conv(B1,d1);
B4d=conv(B4,d4);
[H2,F] = FREQZ(B2d,1,2000,12000);
[H3,F] = FREQZ(B3d,1,2000,12000);
[H5,F] = FREQZ(B5d,1,2000,12000);
[H1,F] = FREQZ(B1d,1,2000,12000);
[H4,F] = FREQZ(B4d,1,2000,12000);
for i=0:24
  d=zeros(1,1+i);
  d(1+i)=1.0;
  B6d=conv(B6,d);
  [H6,F] = FREQZ(B6d,1,2000,12000);
  H=H1+H2+H3+H4+H5+H6+H7+H8;
  I=ones(1810,1);
  E=sum(abs(I-abs(H(1:1810))))
  if E<Ec
    Ec=E;
    In6=i;
  end
end
In6
MATLAB CODE FOR EIGENFILTER DESIGN

wp=0.035*pi;
ws=0.09*pi;
alpha=0.5;
N=32; % this is actually (N-1)/2
for n=0:1:N
    for m=0:1:N
        if n==0&m==0
            A=0;
            B=0;
            C=pi;
            D=ws;
        elseif n==0|m==0
            A=0;
            B=0;
            x=pi;
            C=1/2*sin((-n+m)*x)/(-n+m)+1/2*sin((n+m)*x)/(n+m);
            x=ws;
            D=1/2*sin((-n+m)*x)/(-n+m)+1/2*sin((n+m)*x)/(n+m);
        elseif n==m
            x=wp;
            A=x-2*sin(n*x)/n+(1/2*cos(n*x)*sin(n*x)+1/2*n*x)/n;
            x=0;
            B=x-2*sin(n*x)/n+(1/2*cos(n*x)*sin(n*x)+1/2*n*x)/n;
            x=pi;
            C=(1/2*cos(n*x)*sin(n*x)+1/2*n*x)/n;
            x=ws;
            D=(1/2*cos(n*x)*sin(n*x)+1/2*n*x)/n;
        else
            x=wp;
            A=x-sin(m*x)/m-sin(n*x)/n+1/2*sin((-n+m)*x)/(-n+m)+1/2*sin((n+m)*x)/(n+m);
            x=0;
            B=x-sin(m*x)/m-sin(n*x)/n+1/2*sin((-n+m)*x)/(-n+m)+1/2*sin((n+m)*x)/(n+m);
            x=pi;
            C=1/2*sin((-n+m)*x)/(-n+m)+1/2*sin((n+m)*x)/(n+m);
            x=ws;
            D=1/2*sin((-n+m)*x)/(-n+m)+1/2*sin((n+m)*x)/(n+m);
        end
        P(n+1,m+1)=((1-alpha)/pi)*(A-B)+(alpha/pi)*(C-D);
    end
end
[v,d]=eig(P)
[Y,I]=min(diag(d,0));
b=v(:,I)
L=length(b);
h=ones((2*L)-1,1);
h(L)=b(1);
h(L+1:1:length(h))=b(2:1:L)/2;
h((L-1):-1:1)=b(2:1:L)/2
freqz(h)
Appendix IV

SIMULINK CODE OF HOMOMORPHIC
MULTIPLICATIVE AGC ALGORITHM WITH 8-BAND
IFIR FILTER BANK

**Important Note:** Because of the excessive length of simulinl codes only this simulators code is given.

```plaintext
Model {
    Name "sonic"
    Version 3.00
    SimParamPage "Solver"
    SampleTimeColors off
    InvariantConstants off
    LineColors off
    ShowLineWidths off
    ShowPortDataTypes off
    startTime '0.0'
    stopTime '3.6251'
    SolverMode 'Auto'
    Solver ' ode45'
    RelTol '1e-3'
    AbsTol 'auto'
    Refine '1'
    MaxStep 'auto'
    InitialStep 'auto'
    FixedStep 'auto'
    MaxOrder 5
    OutputOption "RefineOutputTimes"
    OutputTimes '[]'
    LoadExternalInput off
    ExternalInput "[t, u]"
    SaveTime on
    TimeSaveName "tout"
    StateSave off
    StateSaveName "xout"
    SaveOutput on
    OutputSaveName "yout"
    LoadInitialState off
    InitialState "xInitial"
    SaveFinalState off
    FinalStateName "xFinal"
    SaveFormat "Matrix"
    LimitMaxRows off
    MaxRows '1000'
    Decimation '1'
    AlgebraicLoopMsg "warning"
    MinStepSizeMsg "warning"
    UnconnectedInputMsg "warning"
    UnconnectedOutputMsg "warning"
    UnconnectedLineMsg "warning"
    InheritedT2InputMsg "warning"
    IntegerOverflowMsg "warning"
    UnnecessaryDatatypeConvMsg "none"
    Int32ToFloatConvMsg "warning"
}
```

```
SignalLabelMismatchMsg 'none'
ConsistencyChecking 'off'
ZeroCross on
SimulationMode 'normal'
BlockDataTips off
BlockParametersDataTip on
BlockAttributesDataTip off
BlockPortWidthsDataTip off
BlockDescriptionStringDataTip off
BlockMaskParametersDataTip off
ToolBar on
StatusBar on
BrowserShowLibraryLinks off
BrowserLookUnderMask off
OptimizeBlockIOStorage on
BufferReuse on
BooleanDataType off
RTWSystemTargetFile "grt.tlc"
RTWInlineParameters off
RTWRetainRTWFile off
RTWTemplateMakefile "grt_default.tmf"
RTWMakeCommand "make_rtw"
RTWGenerateCodeOnly off
ExtModeMexFile "ext_comm"
ExtModeBatchMode off
ExtModeTrigType "manual"
ExtModeTrigMode "oneshot"
ExtModeTrigPort '1'
ExtModeTrigElement "any"
ExtModeTrigDuration 1000
ExtModeTrigHoldOff 0
ExtModeTrigDelay 0
ExtModeTrigDirection "rising"
ExtModeTrigLevel 0
ExtModeArchiveMode "off"
ExtModeAutoOnOneShot off
ExtModeIncDirWhenArm off
ExtModeAddSufflxToVar off
ExtModeWriteAllDataTows off
ExtModeArmWhenConnect off
Created "Mon Nov 06 14:32:34 2000"
UpdateHistory
"UpdateHistoryNever"
ModifiedByFormat "%<Auto>"
LastModifiedBy "onat"

160
```
Block {  
  BlockType "Delay5"  
  Name "Subsystem"  
  Ports [1. 1. 0. 0. 0]  
  Position [410, 251, 455, 389]  
  SourceBlock "dsptb5sp"  
  SourceType "Delay"  
  N "28*"  
  ic "0*"  
}

Block {  
  BlockType SubSystem  
  Name "Subsystem"  
  Ports [1. 2. 0. 0. 0]  
  Position [95, 239, 130, 286]  
  ShowPortLabels on  
  System {  
    Name "Subsystem"  
    Location [256, 122, 622, 329]  
    Open off  
    ModelBrowserVisibility off  
    ModelBrowserWidth 200  
    ScreenColor "white"  
    PaperOrientation "landscape"  
    PaperPageSize "auto"  
    PaperType "usletter"  
    PaperUnits "inches"  
    ZoomFactor "100"  
    AutoZoom on  
    Block {  
      BlockType "In1"  
      Name "In1"  
      Position [35, 98, 65, 112]  
      Port "1"  
      PortWidth "1*"  
      SampleTime "-1"  
      DataType "auto"  
      SignalType "auto"  
      Interpolate on  
    }  
  }  
  Block {  
    BlockType DiscreteFilter  
    Name "Discrete Filter"  
    Position [115, 47, 175, 83]  
    Numerator "h1"  
    Denominator "[1]*"  
    SampleTime "1/Fs"  
  }  
  Block {  
    BlockType DiscreteFilter  
    Name "Discrete Filter1"  
    Position [115, 122, 175, 158]  
    Numerator "h1c"  
    Denominator "[1]*"  
    SampleTime "1/Fs"  
}

Block {  
  BlockType Outport  
  Name "Out1"  
  Position [220, 58, 250, 72]  
  Port "1"  
  OutputWhenDisabled "held"  
  InitialOutput "[1]*"  
}

Block {  
  BlockType Outport  
  Name "Out2"  
  Position [220, 133, 250, 147]  
  Port "2"  
  OutputWhenDisabled "held"  
  InitialOutput "[1]*"  
}

Line {  
  SrcBlock "In1"  
  SrcPort 1  
  Points [30, 0]  
  Branch {  
    DstBlock "Discrete Filter"  
    DstPort 1  
  }  
  Branch {  
    DstBlock "Discrete Filter1"  
    DstPort 1  
  }  
}

Line {  
  SrcBlock "Discrete Filter1"  
  Filter1"  
  SrcPort 1  
  DstBlock "Out1"  
  DstPort 1  
}

Line {  
  SrcBlock "Discrete Filter1"  
  Filter1"  
  SrcPort 1  
  DstBlock "Out2"  
  DstPort 1  
}

Block {  
  BlockType SubSystem  
  Name "Subsystem"  
  Ports [1. 2. 0. 0. 0]  
  Position [170, 139, 205, 186]  
  ShowPortLabels on  
  System {  
    Name "Subsystem"  
    Location [190, 242, 556, 449]  
    Open off  
    ModelBrowserVisibility off  
    ModelBrowserWidth 200  
    ScreenColor "white"  
    PaperOrientation "landscape"  
    PaperPageSize "auto"  
    PaperType "usletter"  
    PaperUnits "inches"  
    ZoomFactor "100"  
    AutoZoom on  
    Block {  
      BlockType Import  
      Name "In1"  
      Position [35, 98, 65, 112]  
      Port "1"  
      PortWidth "1*"  
      SampleTime "-1"  
      DataType "auto"  
      SignalType "auto"  
      Interpolate on  
    }  
  }  
  Block {  
    BlockType DiscreteFilter  
    Name "Discrete Filter"  
    Position [115, 47, 175, 83]  
    Numerator "h1"  
    Denominator "[1]*"  
    SampleTime "1/Fs"  
  }

163
**Discrete Filter**

**Filter1**

**Numerator** "h2"
**Denominator** "[1]"
**SampleTime** "1/Fs"

**Block**

**Discrete Filter**

**Name** "Discrete Filter1"

**Position** [115, 122, 175, 158]

**Numerator** "h2c"
**Denominator** "[1]"
**SampleTime** "1/Fs"

**Block**

**Discrete Filter**

**Name** "Discrete Filter1"

**Position** [115, 122, 175, 158]

**Numerator** "h2c"
**Denominator** "[1]"
**SampleTime** "1/Fs"

**Block**

**Discrete Filter**

**Name** "Discrete Filter1"

**Position** [115, 122, 175, 158]

**Numerator** "h2c"
**Denominator** "[1]"
**SampleTime** "1/Fs"

**Block**

**Discrete Filter**

**Name** "Discrete Filter1"

**Position** [115, 122, 175, 158]

**Numerator** "h2c"
**Denominator** "[1]"
**SampleTime** "1/Fs"

**Block**

**Discrete Filter**

**Name** "Discrete Filter1"

**Position** [115, 122, 175, 158]

**Numerator** "h2c"
**Denominator** "[1]"
**SampleTime** "1/Fs"

**Block**

**Discrete Filter**

**Name** "Discrete Filter1"

**Position** [115, 122, 175, 158]

**Numerator** "h2c"
**Denominator** "[1]"
**SampleTime** "1/Fs"

**Block**

**Discrete Filter**

**Name** "Discrete Filter1"

**Position** [115, 122, 175, 158]

**Numerator** "h2c"
**Denominator** "[1]"
**SampleTime** "1/Fs"

**Block**

**Discrete Filter**

**Name** "Discrete Filter1"

**Position** [115, 122, 175, 158]

**Numerator** "h2c"
**Denominator** "[1]"
**SampleTime** "1/Fs"
Block {
    BlockType "DiscreteFilter"
    Name "Filter1"
    Port "out1" "held"
    InitialOutput "[]"
    Name "Out1"
    Position [220, 58, 250, 72]
    OutputWhenEnabled "held"
    ![Diagram]
}
Block {
    BlockType "DiscreteFilter"
    Name "Filter1"
    Port "out2" "held"
    InitialOutput "[]"
    Name "Out2"
    Position [220, 133, 250, 147]
    OutputWhenEnabled "held"
    ![Diagram]
}
Block {
    BlockType "SubSystem"
    Name "Subsystem6"
    Ports [1, 2, 0, 0, 0]
    ![Diagram]
}
Block {
    BlockType "Import"
    Name "In1"
    ![Diagram]
}
Block {
    BlockType "Import"
    Name "In2"
    ![Diagram]
}
Block {
    BlockType "SubSystem"
    Name "Subsystem7"
    ![Diagram]
}
Block {
    BlockType "DiscreteFilter"
    Name "Filter"
    ![Diagram]
}
Block {
    BlockType "DiscreteFilter"
    Name "Filter"
    ![Diagram]
}
Port "7"
OutputWhenDisabled "held"
InitialOutput "[1]"

Block {
BlockType Outport
Name "Out8"
Position [510, 438, 540, 452]
Port "8"
OutputWhenDisabled "held"
InitialOutput "[1]"
}

Line {
SrcBlock "In1"
SrcPort 1
DstBlock "Subsystem"
DstPort 1
}

Line {
SrcBlock "Subsystem"
SrcPort 2
Points [20, 0]
DstBlock "Subsystem2"
DstPort 1
}

Line {
SrcBlock "Subsystem"
SrcPort 1
Points [20, 0]
DstBlock "Subsystem1"
DstPort 1
}

Line {
SrcBlock "Subsystem1"
SrcPort 1
Points [25, 0]
DstBlock "Subsystem4"
DstPort 1
}

Line {
SrcBlock "Subsystem1"
SrcPort 2
Points [25, 0]
DstBlock "Subsystem5"
DstPort 1
}

Line {
SrcBlock "Subsystem4"
SrcPort 1
Points [10, 0]
DstBlock "Subsystem3"
DstPort 1
}

Line {
SrcBlock "Subsystem2"
SrcPort 1
Points [25, 0]
DstBlock "Subsystem6"
DstPort 1
}

Line {
SrcBlock "Subsystem2"
SrcPort 2
Points [25, 0]
DstBlock "Subsystem7"
DstPort 1
}

Line {
SrcBlock "Subsystem3"
SrcPort 1
DstBlock "Delay"
}

Line {
SrcBlock "Subsystem4"
SrcPort 2
Points [105, 0]
DstBlock "Delay1"
DstPort 1
}

Line {
SrcBlock "Subsystem5"
SrcPort 1
Points [45, 0; 0, -15]
DstBlock "Delay2"
DstPort 1
}

Line {
SrcBlock "Subsystem5"
SrcPort 2
Points [60, 0; 0, 20]
DstBlock "Delay3"
DstPort 1
}

Line {
SrcBlock "Subsystem6"
SrcPort 1
Points [40, 0; 0, -20]
DstBlock "Delay4"
DstPort 1
}

Line {
SrcBlock "Subsystem6"
SrcPort 2
Points [40, 0; 0, 25]
DstBlock "Delay5"
DstPort 1
}

Line {
SrcBlock "Subsystem7"
SrcPort 1
DstBlock "Out7"
DstPort 1
}

Line {
SrcBlock "Subsystem7"
SrcPort 2
Points [205, 0]
DstBlock "Out8"
DstPort 1
}

Line {
SrcBlock "Delay"
SrcPort 1
DstBlock "Out1"
DstPort 1
}

Line {
SrcBlock "Delay1"
SrcPort 1
DstBlock "Out2"
DstPort 1
}

Line {
SrcBlock "Delay2"
SrcPort 1
DstBlock "Out3"
DstPort 1
}

Line {
SrcBlock "Delay3"
SrcPort 1
}
Block {
    BlockType = Import
    Name = "In1"
    Position = [690, 298, 720, 312]
    BackgroundColor = "cyan"
    Port "3"
    PortWidth = "-1"
    SampleTime = "-1"
    DataType = "auto"
    SignalType = "auto"
    Interpolate on
}

Block {
    BlockType = Import
    Name = "In5"
    Position = [750, 298, 780, 312]
    BackgroundColor = "cyan"
    Port "4"
    PortWidth = "-1"
    SampleTime = "-1"
    DataType = "auto"
    SignalType = "auto"
    Interpolate on
}

Block {
    BlockType = Abs
    Name = "Abs"
    Position = [100, 70, 130, 100]
    BackgroundColor = "red"
}

Block {
    BlockType = Constant
    Name = "Constant"
    Position = [595, 195, 625, 225]
    BackgroundColor = "green"
    Value = "1"
}

Block {
    BlockType = Constant
    Name = "Constant1"
    Position = [795, 350, 825, 380]
    BackgroundColor = "green"
    Value = "330"
}

Block {
    BlockType = SubSystem
    Name = "FastMoving"
    Ports = [1, 1, 0, 0, 0]
    Position = [155, 57, 185, 113]
    BackgroundColor = "orange"
    DropShadow = on
    ShowPortLabels = off
    System "FastMoving"
    Location = [65, 141, 853, 482]
    Open = off
    ModelBrowserVisibility = off
    ModelBrowserWidth = 200
    ScreenColor = "lightBlue"
    PaperOrientation = "landscape"
    PaperPositionMode = "auto"
    PaperType = "usletter"
    PaperUnits = "inches"
    ZoomFactor = "100"
    AutoZoom = on
    Block {
        BlockType = Import
        Name = "In1"
        Position = [75, 93, 105, 107]
        BackgroundColor = "cyan"
        Port = "1"
        PortWidth = "-1"
        SampleTime = "-1"
        DataType = "auto"
        SignalType = "auto"
        Interpolate on
    }
    Block {
        BlockType = Constant
        Name = "Constant"
        Position = [80, 134, 110, 156]
        BackgroundColor = "red"
        Value = "160"
    }
    Block {
        BlockType = DataStoreMemory
        Name = "DataStore\Memory"
        Position = [430, 35, 462, 65]
        BackgroundColor = "orange"
        DataStoreName = "AGC1"
        InitialValue = "0"
    }
    Block {
        BlockType = DataStoreRead
        Name = "DataStore\Read"
        Position = [210, 155, 240, 185]
        BackgroundColor = "orange"
        DataStoreName = "AGC1"
        SampleTime = "1/Fs"
    }
    Block {
        BlockType = DataStoreWrite
        Name = "DataStore\Write"
        Position = [435, 175, 465, 205]
        BackgroundColor = "orange"
        DataStoreName = "AGC1"
        SampleTime = "1/Fs"
    }
    Block {
        BlockType = Reference
        Name = "Integer Delay"
        Position = [200, 67, 245, 103]
        BackgroundColor = "red"
        SourceBlock = "dsb\dsb2/Integer Delay"
        SourceType = "Integer"
        Delay = "160"
        ic = "0"
        frame = off
df = on
        numChans = "1"
    }
    Block {
        BlockType = Product
Name "Product"
Ports [2, 1, 0, 0, 0]
Position [145, 92, 175, 123]
BackgroundColor "green"
Inputs "--" / SaturateOnIntegerOverflow on 
)
Block {
  BlockType Sum
  Name "Sum"
  Ports [2, 1, 0, 0, 0]
  Position [375, 121, 405, 154]
  BackgroundColor "green"
  DropShadow on
  IconShape "rectangular"
  Inputs "--" / SaturateOnIntegerOverflow on 
}
Block {
  BlockType Sum
  Name "Sum"
  Ports [2, 1, 0, 0, 0]
  Position [280, 77, 310, 108]
  BackgroundColor "green"
  DropShadow on
  IconShape "rectangular"
  Inputs "--" / SaturateOnIntegerOverflow on 
}
Block {
  BlockType Outport
  Name "Out1"
  Position [490, 133, 520, 147]
  BackgroundColor "cyan"
  Port "1*
  OutputWhenDisabled "held"
  InitialOutput "0"
}
Line {
  SrcBlock "Constant"
  SrcPort 1
  Points [15, 0]
  DstBlock "Product"
  DstPort 2
}
Line {
  SrcBlock "Init"
  SrcPort 1
  DstBlock "Product"
  DstPort 1
}
Line {
  SrcBlock "Data"
  SrcPort 1
  Points [20, 0]
  DstBlock "Sum1"
  DstPort 2
}
Line {
  SrcBlock "Integer Delay"
  SrcPort 1
  DstBlock "Sum1"
  DstPort 1
}
Line {
  SrcBlock "Sum1"
  SrcPort 1
  Points [45, 0]
  DstBlock "Sum"
  DstPort 1
}
Block {
  BlockType "Product"
  SrcPort 1
  Points [5, 0]
  Branch {
    Points [0, 0]
    DstBlock "Integer Delay"
    DstPort 1
  }
  Branch {
    Points [0, 35]
    DstBlock "Sum"
    DstPort 2
  }
}
Block {
  BlockType "Sum"
  SrcPort 1
  Points [10, 0]
  Branch {
    DstBlock "Data"
    DstPort 1
  }
  Branch {
    DstBlock "Out1"
    DstPort 1
  }
}
Block {
  BlockType Fcn
  Name "Fcn"
  Position [720, 130, 780, 160]
  BackgroundColor "yellow"
  Expr "power(10,((u/20)-5.5))"
}
Block {
  BlockType Fcn
  Name "Fcn1"
  Position [625, 110, 685, 140]
  BackgroundColor "yellow"
  Expr "power(10,((u/20)-5.5))"
}
Block {
  BlockType Logic
  Name "Logical\nOperator2"
  Ports [1, 1, 0, 0, 0]
  Position [450, 144, 480, 176]
  BackgroundColor "yellow"
  Operator "NOT"
  Inputs "1"
}
Block {
  BlockType Math
  Name "Math\nFunction"
  Ports [1, 1, 0, 0, 0]
  Position [960, 75, 990, 105]
  BackgroundColor "red"
  ShowName "off"
}
Operator "log"
OutputSignalType "auto"
}

Block {
    BlockType Math
    Name "Math\nFunction1"
    Ports [1, 1, 0, 0, 0]
    Position [1110, 85, 1140, 115]
    BackgroundColor "red"
    ShowName off
    Operator "exp"
    OutputSignalType "auto"
}

Block {
    BlockType Product
    Name "Product"
    Ports [2, 1, 0, 0, 0]
    Position [800, 72, 830, 103]
    BackgroundColor "yellow"
    Inputs */*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType Product
    Name "Product1"
    Ports [2, 1, 0, 0, 0]
    Position [880, 252, 910, 283]
    BackgroundColor "yellow"
    Inputs */*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType Product
    Name "Product10"
    Ports [2, 1, 0, 0, 0]
    Position [500, 67, 530, 98]
    BackgroundColor "green"
    Inputs */*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType Product
    Name "Product11"
    Ports [2, 1, 0, 0, 0]
    Position [390, 257, 420, 288]
    BackgroundColor "green"
    Inputs */*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType Product
    Name "Product12"
    Ports [2, 1, 0, 0, 0]
    Position [390, 212, 420, 243]
    BackgroundColor "green"
    Inputs */*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType Product
    Name "Product13"
    Ports [2, 1, 0, 0, 0]
    Position [490, 311, 520, 344]
    BackgroundColor "green"
    Inputs */*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType Product
    Name "Product14"
    Ports [2, 1, 0, 0, 0]
    Position [500, 217, 530, 248]
    BackgroundColor "green"
    Inputs */*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType Product
    Name "Product2"
    Ports [2, 1, 0, 0, 0]
    Position [1060, 82, 1090, 113]
    BackgroundColor "yellow"
    Inputs */*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType Product
    Name "Product3"
    Ports [2, 1, 0, 0, 0]
    Position [1175, 77, 1205, 108]
    BackgroundColor "yellow"
    Inputs */*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType RelationalOperator
    Name "Relational\noperator3"
    Position [395, 142, 425, 173]
    BackgroundColor "yellow"
    Operator ">="
}

Block {
    BlockType SubSystem
    Name "SlowMoving"
    Ports [1, 1, 0, 0, 0]
    Position [155, 137, 185, 193]
    BackgroundColor "orange"
    DropShadow on
    ShowPortLabels off
    System {
        Name "SlowMoving"
        Location [65, 141, 853, 482]
        Open off
        ModelBrowserVisibility off
        ModelBrowserWidth 200
        ScreenColor "lightBlue"
        PaperOrientation "landscape"
        PaperPositionMode "auto"
        PaperType "usletter"
        PaperUnits "inches"
        ZoomFactor "100"
        AutoZoom on
        Block {
            BlockType Import
            Name "In1"
        }
    }
}
Position [75, 93, 175, 123]
BackgroundColor "cyan"
Port "1" "1"
PortWidth "-1"
SampleTime "-1"
DataType "auto"
SignalType "auto"
Interpolate on

Block {
  BlockType Constant
  Name "Constant"
  Position [80, 134, 180, 156]
  BackgroundColor "red"
  Value "3520"
}

Block (DataStoreMemory)
Name "Data"
StoreLocation Name "AGCSI"
InitialValue "0"

Block (DataStoreRead)
Name "Data"
StoreLocation Name "AGCSI"
SampleTime "1/Fs"

Block (DataStoreWrite)
Name "Data"
StoreLocation Name "AGCSI"
SampleTime "1/Fs"

Block (Reference)
Name "Integer Delay"
Ports [1, 1, 0, 0, 0]
Position [200, 67, 245, 103]
BackgroundColor "red"
Source Block "dspbdsp2/Integer Delay"
SourceType "Integer"

Delay
  delay "3520"
  ic "0"
  frame off
  df on
  numChans "1"
}

Block (Product)
Name "Product"
Ports [2, 1, 0, 0, 0]
Position [145, 92, 175, 123]
BackgroundColor "green"
Inputs "1"
SaturateOnIntegerOverflow on

Block {
  BlockType Sum
  Name "Sum"
  Ports [2, 1, 0, 0, 0]
  Position [375, 121, 310, 108]
  BackgroundColor "green"
  DropShadow on
  IconShape "rectangular"
  Inputs "1"
  SaturateOnIntegerOverflow on
}

Block (DataStoreMemory)
Name "Data"
StoreLocation Name "Sum"

Block (DataStoreWrite)
Name "Data"
StoreLocation Name "Sum"
DstBlock "Product"
DstPort 2

Block (DataStoreRead)
Name "Data"
StoreLocation Name "Product"
DstPort 1

Line {
  SrcBlock "Constant"
  SrcPort 1
  Points [15.0]
  DstBlock "Product"
  DstPort 2
}

Line {
  SrcBlock "In1"
  SrcPort 1
  DstBlock "Product"
  DstPort 1
}

Line (DataStoreRead)
Name "Data"

Line (DataStoreWrite)
Name "Data"

174
Name "dB Conv*
Location [12, 74, 478, 310, 100]
Open off
ModelBrowserVisibility off
ModelBrowserWidth 200
ScreenColor "lightblue"
PaperOrientation "landscape"

PaperPositionMode "auto"
PaperType "usletter"
PaperUnits "inches"
ZoomFactor "100"
AutoZoom on

Block {
  BlockType Import
  Name "In1"
  Position [25, 68, 55, 82]
  BackgroundColor "cyan"
  Port "1"
  PortWidth "1"
  SampleTime "-1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}

Block {
  BlockType Constant
  Name "Constant1"
  Position [100, 80, 120, 100]
  BackgroundColor "darkGreen"
  Value "0"
}

Block {
  BlockType Fcn
  Name "Fcn"
  Position [185, 165, 245, 195]
  BackgroundColor "yellow"
  Expr "20*log10(u/(3.1623*power(10,-6)))"
}

Block {
  BlockType RelationalOperator
  Name "Relational\nOperator"
  Position [145, 67, 175, 98]
  BackgroundColor "yellow"
  Operator "<="
}

Block {
  BlockType Sum
  Name "Sum"
  Ports [2, 1, 0, 0, 0]
  Position [120, 162, 150, 193]

  BackgroundColor "green"
  IconShape "rectangular"
  Inputs "++
  SaturateOnIntegerOverflow on"
}

Block {
  BlockType Switch
  Name "Switch"
  BackgroundColor "orange"
  Threshold "1"
}

Block {
  BlockType Outport
  Name "Out1"
  Position [360, 78, 390, 92]
  BackgroundColor "cyan"
  Port "1"
  OutputWhenDisabled "held"
  InitialOutput "[]"
}

Line {
  SrcBlock "Fcn"
  SrcPort 1
  Points [15, 0]
  DstBlock "Switch"
  DstPort 3
}

Line {
  SrcBlock "In1"
  SrcPort 1
  Points [10, 0]
  Branch {
    Points [0, -45; 195, 0]
    DstBlock "Switch"
    DstPort 1
  }
  Branch {
    DstBlock "Relational\nOperator"
    DstPort 1
  }
  Branch {
    Points [0, 110]
    DstBlock "Sum"
    DstPort 2
  }
}

Line {
  SrcBlock "Constant1"
  SrcPort 1
  DstBlock "Relational\nOperator"
  DstPort 2
}

Line {
  SrcBlock "Relational\nOperator"
  SrcPort 1
  Points [10, 0]
  Branch {
    DstBlock "Switch"
    DstPort 2
  }
  Branch {
    Points [0, 50; -105, 0]
    DstBlock "Sum"
    DstPort 1
  }
}

Line {
  SrcBlock "Sum"
  SrcPort 1
  DstBlock "Fcn"
  DstPort 1
}

Line {
Block {
    BlockType    "Import"
    Name         "In3"
    Position     [710, 298, 740, 312]
    BackgroundColor "cyan"
    Port         "-1"
    PortWidth    "-1"
    SampleTime   "-1"
    DataType     "auto"
    SignalType   "auto"
    Interpolate  on
}
Block {
    BlockType    "Import"
    Name         "In4"
    Position     [805, 298, 835, 312]
    BackgroundColor "cyan"
    Port         "3"
    PortWidth    "-1"
    SampleTime   "-1"
    DataType     "auto"
    SignalType   "auto"
    Interpolate  on
}
Block {
    BlockType    "Import"
    Name         "In5"
    Position     [865, 298, 895, 312]
    BackgroundColor "cyan"
    Port         "4"
    PortWidth    "-1"
    SampleTime   "-1"
    DataType     "auto"
    SignalType   "auto"
    Interpolate  on
}
Block {
    BlockType    "Abs"
    Name         "Abs"
    Position     [215, 70, 245, 100]
    BackgroundColor "red"
}
Block {
    BlockType    "Constant"
    Name         "Constant"
    Position     [710, 195, 740, 225]
    BackgroundColor "green"
    Value        "1"
}
Block {
    BlockType    "Constant"
    Name         "Constant1"
    Position     [910, 350, 940, 380]
    BackgroundColor "green"
    Value        "330"
}

Block {
    BlockType    "SubSystem"
    Name         "FastMoving"
    Ports        [1, 1, 0, 0, 0]
    Position     [270, 57, 300, 113]
    BackgroundColor "orange"
    DropShadow   on
    ShowPortLabels off
    System {
        Name         "FastMoving"
        Location     [65, 141, 853, 482]
        Open         off
        ModelBrowserVisibility off
        ModelBrowserWidth  200
        ScreenColor    "lightBlue"
        PaperOrientation "landscape"
        PaperPositionMode "auto"
        PaperType     "usletter"
        PaperUnits    "inches"
        ZoomFactor    "100"
        AutoZoom      on
    }
    Block {
        BlockType    "Import"
        Name         "In1"
        Position     [75, 93, 105, 107]
        BackgroundColor "cyan"
        Port         "-1"
        PortWidth    "-1"
        SampleTime   "-1"
        DataType     "auto"
        SignalType   "auto"
        Interpolate  on
    }
    Block {
        BlockType    "Constant"
        Name         "Constant"
        Position     [80, 134, 110, 156]
        BackgroundColor "red"
        Value        "160"
    }
    Block {
        BlockType    "DataStoreMemory"
        Name         "DataStore\nMemory"
        Store\nName    "AGC2"
        Value        "o"
    }
    Block {
        BlockType    "DataStoreRead"
        Name         "DataStore\nRead"
        Position     [210, 155, 240, 185]
        BackgroundColor "orange"
        DataStoreName "AGC2"
        SampleTime   "1/fs"
    }
    Block {
        BlockType    "DataStoreWrite"
BackgroundColor "yellow"
Operator "=="
}
Block {
    BlockType SubSystem
    Name "SlowMoving"
    Ports [1, 1, 0, 0, 0]
    Position [270, 137, 300, 193]
    BackgroundColor "orange"
    DropShadow on
    ShowPortLabels off
    System {
        Name "SlowMoving"
        Location [67, 139, 855, 480]
        Open off
        ModelBrowserVisibility off
        ModelBrowserWidth 200
        ScreenColor "lightBlue"
        PaperOrientation "landscape"
        PaperPositionMode "auto"
        PaperType "usletter"
        PaperUnits "inches"
        ZoomFactor "100"
        AutoZoom on
    }
    Block {
        BlockType Import
        Name "In1"
        Position [75, 93, 105, 107]
        BackgroundColor "cyan"
        Port "1"
        PortWidth "-1"
        SampleTime "-1"
        DataType "auto"
        SignalType "auto"
        Interpolate on
    }
    Block {
        BlockType Constant
        Name "Constant"
        Position [80, 134, 110, 156]
        BackgroundColor "red"
        Value "3520"
    }
}
Block {
    BlockType DataStoreMemory
    Name "Data"
    Store\nMemory
    Position [430, 35, 462, 65]
    BackgroundColor "orange"
    DataStoreName "AGCS2"
    InitialValue "0"
}
Block {
    BlockType DataStoreRead
    Name "Data"
    Store\nRead
    Position [210, 155, 240, 185]
    BackgroundColor "orange"
    DataStoreName "AGCS2"
    SampleTime "1/Fs"
}
Block {
    BlockType DataStoreWrite
    Name "Data"
    Store\nWrite
    Position [435, 175, 465, 205]
    BackgroundColor "orange"
    DataStoreName "AGCS2"
    SampleTime "1/Fs"
}
Block {
    BlockType Reference
    Name "Integer Delay"
    Ports [1, 1, 0, 0, 0]
    Position [200, 67, 245, 103]
    BackgroundColor "red"
    SourceBlock "dspbdsp2/Integer Delay"
    SourceType "Integer"
    Delay delay "3520"
    ic 0 frame on
    df 0 numChans 1
}
Block {
    BlockType Product
    Name "Product"
    Ports [2, 1, 0, 0, 0]
    Position [145, 92, 175, 123]
    BackgroundColor "green"
    Inputs ///
    SaturateOnIntegerOverflow on
}
Block {
    BlockType Sum
    Name "Sum"
    Ports [2, 1, 0, 0, 0]
    Position [375, 121, 405, 154]
    BackgroundColor "green"
    DropShadow on
    IconShape "rectangular"
    Inputs ///
    SaturateOnIntegerOverflow on
}
Block {
    BlockType Outport
    Name "Out1"
    Position [490, 133, 520, 147]
    BackgroundColor "cyan"
    Port "1"
OutputWhenDisabled "held"
InitialOutput "0"

Block {
  SrcBlock "Constant"
  SrcPort 1
  Points [15, 0]
  DstBlock "Product"
  DstPort 2
}

Block {
  SrcBlock "Inl"
  SrcPort 1
  DstBlock "Product"
  DstPort 1
}

Block {
  SrcBlock "Data"
  SrcPort 1
  Points [20, 0]
  DstBlock "Sum1"
  DstPort 2
}

Block {
  SrcBlock "Integer Delay"
  SrcPort 1
  Points [45, 0]
  DstBlock "Sum"
  DstPort 1
}

Block {
  SrcBlock "Sum1"
  SrcPort 1
  Points [45, 0]
  DstBlock "Integer Delay"
  DstPort 1
}

Branch {
  Points [0, 0]
  DstBlock "Sum"
  DstPort 1
}

Branch {
  Points [0, 35]
  DstBlock "Sum"
  DstPort 2
}

Branch {
  SrcBlock "Sum"
  SrcPort 1
  Points [10, 0]
  DstBlock "Data"
  DstPort 1
}

Branch {
  DstBlock "Outl"
  DstPort 1
}

}

Block {
  BlockType Sum
  Name "Sum"
  Ports [2, 1, 0, 0, 0]
  Position [232, 232, 1055, 203, 1075]
  Background "orange"
  ShowName off
  IconShape "rectangular"
  Inputs "--"
  SaturateOnIntegerOverflow on
}

Block {
  BlockType Sum
  Name "Sum1"
  Ports [2, 1, 0, 0, 0]
  Position [450, 262, 480, 293]
  Background "green"
  IconShape "rectangular"
  Inputs "--"
  SaturateOnIntegerOverflow on
}

Block {
  BlockType Sum
  Name "Sum2"
  Ports [2, 1, 0, 0, 0]
  Position [555, 252, 585, 283]
  Background "green"
  IconShape "rectangular"
  Inputs "--"
  SaturateOnIntegerOverflow on
}

Block {
  BlockType Sum
  Name "Sum3"
  Ports [2, 1, 0, 0, 0]
  Position [1130, 213, 1150, 242]
  Background "orange"
  ShowName off
  IconShape "rectangular"
  Inputs "--"
  SaturateOnIntegerOverflow on
}

Block {
  BlockType Sum
  Name "Sum4"
  Ports [2, 1, 0, 0, 0]
  Position [1075, 293, 1095, 322]
  Background "orange"
  ShowName off

185
InitialOutput = "0"

Line {
    SrcBlock = "In1"
    SrcPort = 1
    Points = [10, 0]
    Branch {
        Points = [0, -45; 195, 0]
        DstBlock = "Switch"
        DstPort = 1
        Branch {
            Points = [0, 110]
            DstBlock = "Sum"
            DstPort = 2
        }
    }
    Line {
        SrcBlock = "Constant1"
        SrcPort = 1
        DstBlock = "Switch"
        DstPort = 2
        Branch {
            Points = [0, 50; -105, 0]
            DstBlock = "Sum"
            DstPort = 1
        }
    }
    Line {
        SrcBlock = "Sum"
        SrcPort = 1
        DstBlock = "Pcm"
        DstPort = 1
    }
    Line {
        SrcBlock = "Switch"
        SrcPort = 1
        DstBlock = "Out1"
        DstPort = 1
    }
}

Block {
    BlockType = Constant
    Name = "effect"
    Position = [370, 210, 390, 230]
    BackgroundColor = "red"
    Value = "6"
}

Block {
    BlockType = Outport
    Name = "Out1"
    Position = [1415, 88, 1445, 102]
    BackgroundColor = "cyan"
    Port = "1"
    OutputWhenDisabled = "held"
}
SrcBlock = "Sum13"
SrcPort = 1
DstBlock = "Product13"
DstPort = 1

Branch {
    Points [5, 0]
    Branch {
        Points [-10, 0, 0, 135]
        Branch {
            DstBlock = "Sum12"
            DstPort = 1
        }
    }
}

Line {
    SrcBlock = "Product10"
    SrcPort = 1
    Points [10, 0]
    DstBlock = "Sum8"
    DstPort = 1
}

Branch {
    Points [50, 0]
    DstBlock = "Relation1\nOperator3"
    DstPort = 1
}

Line {
    SrcBlock = "Sum12"
    SrcPort = 1
    Points [0, 0]
    Branch {
        Points [0, 0]
        DstBlock = "Product11"
        DstPort = 2
    }
    Branch {
        Points [0, 45]
        DstBlock = "Sum14"
        DstPort = 1
    }
}

Line {
    SrcBlock = "effect"
    SrcPort = 1
    Points [0, 0]
    Branch {
        DstBlock = "Product11"
        DstPort = 1
    }
    Branch {
        Points [0, 120]
        DstBlock = "Sum14"
        DstPort = 2
    }
}

Line {
    SrcBlock = "Sum14"
    SrcPort = 1
    DstBlock = "Product13"
    DstPort = 2
}

Line {
    SrcBlock = "Product13"
    SrcPort = 1
    Points [0, -65; -40, 0]
    DstBlock = "Product14"
    DstPort = 2
}

Line {
    SrcBlock = "dB Conv"
    SrcPort = 1
    Points [45, 0; 0, 20; 40, 0]
    Branch {
        Points [0, -25]
        DstBlock = "Product10"
        DstPort = 1
    }
}

Line {
    SrcBlock = "FastMoving"
    SrcPort = 1
    Points [5, 0]
    DstBlock = "dB Conv"
    DstPort = 1
}

Line {
    SrcBlock = "SlowMoving"
    SrcPort = 1
    Points [5, 0]
    DstBlock = "dB Conv.-LF"
    DstPort = 1
}

Line {
    SrcBlock = "Pcn1"
    SrcPort = 1
    Points [25, 0; 0, -45]
    DstBlock = "Product"
    DstPort = 1
}

Line {
    SrcBlock = "Sum8"
    SrcPort = 1
}

190
SaturateOnIntegerOverflow on
}
Block {
    BlockType Sum
    Name "Sum"
    Ports [2, 1, 0, 0, 0]
    Position [375, 121.
    BackgroundColor "green"
    DropShadow on
    IconShape "rectangular"
    Inputs "--" SaturateOnIntegerOverflow on
}
Block {
    BlockType Sum
    Name "Sum1"
    Ports [2, 1, 0, 0, 0]
    Position [280, 77.
    BackgroundColor "green"
    DropShadow on
    IconShape "rectangular"
    Inputs "--" SaturateOnIntegerOverflow on
}
Block {
    BlockType Outport
    Name "Out1"
    Position [490, 133,
    BackgroundColor "cyan"
    Port "1"
    OutputWhenDisabled "held"
    InitialOutput "0"
}
Line {
    SrcBlock "Constant"
    SrcPort 1
    Points [15, 0]
    DstBlock "Product"
    DstPort 2
}
Line {
    SrcBlock "In1"
    SrcPort 1
    DstBlock "Product"
    DstPort 1
}
Line {
    SrcBlock "Data"
    SrcPort 1
    Points [20, 0]
    DstBlock "Sum1"
    DstPort 2
}
Line {
    SrcBlock "Integer Delay"
    SrcPort 1
    DstBlock "Sum1"
    DstPort 1
}
Line {
    SrcBlock "Sum1"
    SrcPort 1
    Points [45, 0]
    DstBlock "Sum"
    DstPort 1
}
Line {
    SrcBlock "Product"
    SrcPort 1
    Points [5, 0]
    DstBlock "Integer Delay"
    DstPort 1
}
Line {
    SrcBlock "Sum"
    SrcPort 1
    Points [10, 0]
    DstBlock "Data"
    DstPort 1
}
Line {
    SrcBlock "Out1"
    SrcPort 1
    DstBlock "Data"
    DstPort 1
}
Line {
    SrcBlock "Constant"
    SrcPort 1
    Points [15, 0]
    DstBlock "Product"
    DstPort 2
}
Line {
    SrcBlock "In1"
    SrcPort 1
    DstBlock "Product"
    DstPort 1
}
Line {
    SrcBlock "Data"
    SrcPort 1
    Points [20, 0]
    DstBlock "Sum1"
    DstPort 2
}
Line {
    SrcBlock "Integer Delay"
    SrcPort 1
    DstBlock "Sum1"
    DstPort 1
}
Line {
    SrcBlock "Sum1"
    SrcPort 1
    Points [45, 0]
    DstBlock "Sum"
    DstPort 1
}
Line {
    SrcBlock "Product"
    SrcPort 1
    Points [5, 0]
    DstBlock "Integer Delay"
    DstPort 1
}
Line {
    SrcBlock "Sum"
    SrcPort 1
    Points [10, 0]
    DstBlock "Data"
    DstPort 1
}
Line {
    SrcBlock "Out1"
    SrcPort 1
    DstBlock "Data"
    DstPort 1
}
Line {
    SrcBlock "Constant"
    SrcPort 1
    Points [15, 0]
    DstBlock "Product"
    DstPort 2
}
Line {
    SrcBlock "In1"
    SrcPort 1
    DstBlock "Product"
    DstPort 1
}
Line {
    SrcBlock "Data"
    SrcPort 1
    Points [20, 0]
    DstBlock "Sum1"
    DstPort 2
}
Line {
    SrcBlock "Integer Delay"
    SrcPort 1
    DstBlock "Sum1"
    DstPort 1
}
Line {
    SrcBlock "Sum1"
    SrcPort 1
    Points [45, 0]
    DstBlock "Sum"
    DstPort 1
}
Line {
    SrcBlock "Product"
    SrcPort 1
    Points [5, 0]
    DstBlock "Integer Delay"
    DstPort 1
}
Line {
    SrcBlock "Sum"
    SrcPort 1
    Points [10, 0]
    DstBlock "Data"
    DstPort 1
}
Line {
    SrcBlock "Out1"
    SrcPort 1
    DstBlock "Data"
    DstPort 1
}
Line {
    SrcBlock "Constant"
    SrcPort 1
    Points [15, 0]
    DstBlock "Product"
    DstPort 2
}
Line {
    SrcBlock "In1"
    SrcPort 1
    DstBlock "Product"
    DstPort 1
}
Line {
    SrcBlock "Data"
    SrcPort 1
    Points [20, 0]
    DstBlock "Sum1"
    DstPort 2
}
Line {
    SrcBlock "Integer Delay"
    SrcPort 1
    DstBlock "Sum1"
    DstPort 1
}
Line {
    SrcBlock "Sum1"
    SrcPort 1
    Points [45, 0]
    DstBlock "Sum"
    DstPort 1
}
Threshold "1"

Block {
  BlockType Outport
  Name "Out1"
  Position [360, 78, 390, 92]
  BackgroundColor "cyan"
  Port "1"
  OutputWhenDisabled "held"
  InitialOutput "[]"
}

Line {
  SrcBlock "Fcn"
  SrcPort 1
  Points [15, 0]
  DstBlock "Switch"
  DstPort 3
}

Line {
  SrcBlock "Inl"
  SrcPort 1
  Points [10, 0]
  Branch {
    Points [0, -45; 195, 0]
    DstBlock "Switch"
    DstPort 1
  }
  Branch {
    DstBlock "Relational\nOperator"
    DstPort 1
  }
  Branch {
    Points [0, 110]
    DstBlock "Sum"
    DstPort 2
  }
}

Line {
  SrcBlock "Constant1"
  SrcPort 1
  DstBlock "Relational\nOperator"
  DstPort 2
}

Line {
  SrcBlock "Relational\nOperator"
  SrcPort 1
  Points [30, 0]
  Branch {
    DstBlock "Switch"
    DstPort 2
  }
  Branch {
    Points [0, 50; -105, 0]
    DstBlock "Sum"
    DstPort 1
  }
}

Line {
  SrcBlock "Sum"
  SrcPort 1
  DstBlock "Fcn"
  DstPort 1
}

Line {
  SrcBlock "Switch"
  SrcPort 1
  DstBlock "Out1"

  DstPort 1
}

Block {
  BlockType SubSystem
  Name "dB Conv.-LF"
  Ports [1, 1, 0, 0, 0, 0]
  Position [325, 158, 350, 192]
  BackgroundColor "darkGreen"
  DropShadow on
  ShowPortLabels off
  System {
    Name "dB Conv.-LF"
    Location [152, 320, 618, 645]
    Open off
    ModelBrowserVisibility off
    ModelBrowserWidth 200
    ScreenColor "lightBlue"
    PaperOrientation "landscape"
    PaperPositionMode "auto"
    PaperType "usletter"
    PaperUnits "inches"
    ZoomFactor "100"
    AutoZoom on
  }
  Block {
    BlockType Inport
    Name "Inl"
    Position [25, 68, 55, 82]
    BackgroundColor "cyan"
    Port "1"
    PortWidth "-1"
    SampleTime "-1"
    DataType "auto"
    SignalType "auto"
    Interpolate on
  }
  Block {
    BlockType Constant
    Name "Constant1"
    Position [25, 68, 55, 82]
    BackgroundColor "cyan"
    Value "0"
  }
  Block {
    BlockType Fcn
    Name "Fcn"
    Position [185, 165, 245, 195]
    BackgroundColor "yellow"
    Expr "20*log10(u/(3.1623*power(10,-6)))"
  }
  Block {
    BlockType RelationalOperator
    Name "Relational\nOperator"
    Position [145, 67, 175, 98]
    BackgroundColor "yellow"
    Operator "<="
  }
  Block {
    BlockType Sum
  }
}
<table>
<thead>
<tr>
<th>Name</th>
<th>&quot;Sum&quot;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ports</td>
<td>[2, 1, 0, 0, 0]</td>
</tr>
<tr>
<td>Position</td>
<td>[120, 162, 150, 193]</td>
</tr>
<tr>
<td>BackgroundColor</td>
<td>&quot;green&quot;</td>
</tr>
<tr>
<td>IconShape</td>
<td>&quot;rectangular&quot;</td>
</tr>
<tr>
<td>Inputs</td>
<td>&quot;==='&quot;</td>
</tr>
<tr>
<td>SaturateOnIntegerOverflow</td>
<td>on</td>
</tr>
<tr>
<td>Block</td>
<td></td>
</tr>
<tr>
<td>BlockType</td>
<td>Switch</td>
</tr>
<tr>
<td>Name</td>
<td>&quot;Switch&quot;</td>
</tr>
<tr>
<td>Position</td>
<td>[280, 70, 310, 100]</td>
</tr>
<tr>
<td>BackgroundColor</td>
<td>&quot;orange&quot;</td>
</tr>
<tr>
<td>Threshold</td>
<td>&quot;1&quot;</td>
</tr>
<tr>
<td>Block</td>
<td></td>
</tr>
<tr>
<td>BlockType</td>
<td>Outport</td>
</tr>
<tr>
<td>Name</td>
<td>&quot;Out1&quot;</td>
</tr>
<tr>
<td>Position</td>
<td>[360, 78, 390, 92]</td>
</tr>
<tr>
<td>BackgroundColor</td>
<td>&quot;cyan&quot;</td>
</tr>
<tr>
<td>Port</td>
<td>&quot;1&quot;</td>
</tr>
<tr>
<td>OutputWhenDisabled</td>
<td>&quot;held&quot;</td>
</tr>
<tr>
<td>InitialOutput</td>
<td>&quot;[]&quot;</td>
</tr>
<tr>
<td>Line</td>
<td></td>
</tr>
<tr>
<td>SrcBlock</td>
<td>&quot;Fcn&quot;</td>
</tr>
<tr>
<td>SrcPort</td>
<td>1</td>
</tr>
<tr>
<td>Points</td>
<td>[15, 0]</td>
</tr>
<tr>
<td>DstBlock</td>
<td>&quot;Switch&quot;</td>
</tr>
<tr>
<td>DstPort</td>
<td>3</td>
</tr>
<tr>
<td>Line</td>
<td></td>
</tr>
<tr>
<td>SrcBlock</td>
<td>&quot;In1&quot;</td>
</tr>
<tr>
<td>SrcPort</td>
<td>1</td>
</tr>
<tr>
<td>Points</td>
<td>[10, 0]</td>
</tr>
<tr>
<td>Branch</td>
<td></td>
</tr>
<tr>
<td>Points</td>
<td>[0, -45; 195, 0]</td>
</tr>
<tr>
<td>DstBlock</td>
<td>&quot;Switch&quot;</td>
</tr>
<tr>
<td>DstPort</td>
<td>1</td>
</tr>
<tr>
<td>Branch</td>
<td></td>
</tr>
<tr>
<td>DstBlock</td>
<td>&quot;Relational\nOperator&quot;</td>
</tr>
<tr>
<td>DstPort</td>
<td>1</td>
</tr>
<tr>
<td>Branch</td>
<td></td>
</tr>
<tr>
<td>Points</td>
<td>[0, 110]</td>
</tr>
<tr>
<td>DstBlock</td>
<td>&quot;Sum&quot;</td>
</tr>
<tr>
<td>DstPort</td>
<td>2</td>
</tr>
<tr>
<td>Line</td>
<td></td>
</tr>
<tr>
<td>SrcBlock</td>
<td>&quot;Constant1&quot;</td>
</tr>
<tr>
<td>SrcPort</td>
<td>1</td>
</tr>
<tr>
<td>DstBlock</td>
<td>&quot;Relational\nOperator&quot;</td>
</tr>
<tr>
<td>DstPort</td>
<td>2</td>
</tr>
<tr>
<td>Line</td>
<td></td>
</tr>
<tr>
<td>SrcBlock</td>
<td>&quot;Relational\nOperator&quot;</td>
</tr>
<tr>
<td>SrcPort</td>
<td>1</td>
</tr>
<tr>
<td>Points</td>
<td>[30, 0]</td>
</tr>
<tr>
<td>Branch</td>
<td></td>
</tr>
<tr>
<td>DstBlock</td>
<td>&quot;Switch&quot;</td>
</tr>
<tr>
<td>DstPort</td>
<td>2</td>
</tr>
</tbody>
</table>
BlockType  SubSystem
Name       "FastMoving"
Ports      [1, 1, 0, 0, 0]
Position   [270, 57, 300, 113]
BackgroundColor "orange"
DropShadow  on
ShowPortLabels off
System {
  Name    "FastMoving"
  Location [71, 135, 859,
476]  Open    off
  ModelBrowserVisibility off
  ModelBrowserWidth  200
  ScreenColor "lightBlue"
  PaperOrientation "landscape"
  PaperPositionMode "auto"
  PaperType "usletter"
  PaperUnits "inches"
  ZoomFactor "100"
  AutoZoom   on
  Block {
    BlockType  Import
    Name      "In"
    Position  [75, 93,
105, 107]
    BackgroundColor "cyan"
    Port      "1"
    PortWidth "1"
    SampleTime "-1"
    DataType  "auto"
    SignalType "auto"
    Interpolate on
  }
  Block {
    BlockType  Constant
    Name      "Constant"
    Position  [80, 134,
110, 156]
    BackgroundColor "red"
    Value     "160"
  }
  Block {
    BlockType  DataStoreMemory
    Name      "DataStore\nMemory"
    Store\nMemory" Position  [430, 35,
462, 65]
    BackgroundColor "orange"
    DataStoreName "AGC4"
    InitialValue "0"
  }
  Block {
    BlockType  DataStoreRead
    Name      "DataStore\nRead"
    Store\nRead" Position  [210, 155,
240, 185]
    BackgroundColor "orange"
    DataStoreName "AGC4"
    SampleTime "1/Fs"
  }
  Block {
    BlockType  DataStoreWrite
    Name      "DataStore\nWrite"
    Store\nWrite" Name    "Data"
  }
  Position  [435, 175,
465, 205]
  BackgroundColor "orange"
  DataStoreName "AGC4"
  SampleTime "1/Fs"
}
Block {
  BlockType
  Reference {
    Name    "Integer Delay"
    Ports   [1, 1, 0, 0, 0]
    Position [200, 67,
245, 103]
    BackgroundColor "red"
    SourceBlock "dspbdsp2/Integer Delay"
    SourceType "Integer"
    Delay {
      delay  "160"
      int  0
      frame  off
      df   on
      numChans "1"
    }
    Block {
      BlockType  Product
      Name      "Product"
      Ports     [2, 1, 0, 0, 0]
      Position  [145, 92,
175, 123]
      BackgroundColor "green"
      Inputs "-/-"
      SaturateOnIntegerOverflow on
    }
    Block {
      BlockType  Sum
      Name      "Sum"
      Ports     [2, 1, 0, 0, 0]
      Position  [375, 121,
405, 154]
      BackgroundColor "green"
      DropShadow on
      IconShape "rectangular"
      Inputs "-/-"
      SaturateOnIntegerOverflow on
    }
    Block {
      BlockType  Sum
      Name      "Sum1"
      Ports     [2, 1, 0, 0, 0]
      Position  [280, 77,
310, 108]
      BackgroundColor "green"
      DropShadow on
      IconShape "rectangular"
      Inputs "-/-"
      SaturateOnIntegerOverflow on
    }
    Block {
      BlockType  Outport
      Name      "Out1"
      Position  [490, 133,
520, 147]
      BackgroundColor "cyan"
      Port     "1"
      OutputWhenDisabled "held"
      InitialOutput "0"
    }
    Line {
202
BlockType Fcn
Name "Fcn"
Position [185, 165, 245, 195]
BackgroundColor "yellow"
Expr "20*\log10(u/(3.1623*power(10, -6)))"
}
Block {
BlockType RelationalOperator
Name "Relational\nOperator"
Position [145, 67, 175, 98]
BackgroundColor "yellow"
Operator "<="
}
Block {
BlockType Sum
Name "Sum"
Ports [2, 1, 0, 0, 0]
Position [120, 162, 150, 193]
BackgroundColor "green"
IconShape "rectangular"
Inputs "-+
SaturateOnIntegerOverflow on
}
Block {
BlockType Switch
Name "Switch"
Position [280, 70, 310, 100]
BackgroundColor "orange"
Threshold "1"
}
Block {
BlockType Outport
Name "Out1"
Position [360, 78, 390, 92]
BackgroundColor "cyan"
Port "1"
OutputWhenDisabled "held"
InitialOutput "[1]"
}
Line {
SrcBlock "Fcn"
SrcPort 1
Points [15, 0]
DstBlock "Switch"
DstPort 3
}
Line {
SrcBlock "In1"
SrcPort 1
Points [10, 0]
Branch {
Points [0, -45; 195, 0]
DstBlock "Switch"
DstPort 1
}
Branch {
DstBlock "Relational\nOperator"
DstPort 1
}
Branch {
Points [0, 110]

DstBlock "Sum"
DstPort 2
}
Line {
SrcBlock "Constant1"
SrcPort 1
DstBlock "Relational\nOperator"
DstPort 2
}
Line {
SrcBlock "Relational\nOperator"
SrcPort 1
Points [30, 0]
Branch {
DstBlock "Switch"
DstPort 2
}
Branch {
Points [0, 50; -105, 0]
DstBlock "Sum"
DstPort 1
}
Line {
SrcBlock "Sum"
SrcPort 1
DstBlock "Fcn"
DstPort 1
}
Line {
SrcBlock "Switch"
SrcPort 1
DstBlock "Out1"
DstPort 1
}

Block {
BlockType SubSystem
Name "dB Conv.-LF"
Ports [1, 1, 0, 0, 0]
Position [325, 158, 350, 192]
BackgroundColor "darkGreen"
DropShadow on
ShowPortLabels off
System {
Name "dB Conv.-LF"
Location [152, 320, 618, 645]
Open off
ModelBrowserVisibility off
ModelBrowserWidth 200
ScreenColor "lightBlue"
PaperOrientation "landscape"
PaperPositionMode "auto"
PaperType "usletter"
PaperUnits "inches"
ZoomFactor "100"
AutoZoom on
Block {
BlockType In1
Name "In1"
Position [25, 68, 55, 82]
BackgroundColor "cyan"
Port "1"
SignalType  "auto"
Interpolate on

) Block {
  BlockType Import
  Name "In5"
  Position [865, 298, 895, 312]
  BackgroundColor "cyan"
  Port "4"
  PortWidth "+1"
  SampleTime "+1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}

) Block {
  BlockType Abs
  Name "Abs"
  Position [215, 70, 245, 100]
  BackgroundColor "red"
}

) Block {
  BlockType Constant
  Name "Constant1"
  Position [710, 195, 740, 225]
  BackgroundColor "green"
  Value "1"
}

) Block {
  BlockType Constant
  Name "Constant1"
  Position [710, 195, 740, 225]
  BackgroundColor "green"
  Value "330"
}

) Block {
  BlockType SubSystem
  Name "FastMoving"
  Ports [1, 1, 0, 0, 0]
  Position [270, 57, 300, 113]
  BackgroundColor "orange"
  DropShadow off
  ShowPortLabels off
  System {
    Name "FastMoving"
    Location [73, 133, 861, 474]
    Open off
    ModelBrowserVisibility off
    ModelBrowserWidth 200
    ScreenColor "lightBlue"
    PaperOrientation "landscape"
    PaperPositionMode "auto"
    PaperType "usletter"
    PaperUnits "inches"
    ZoomFactor "100"
    AutoZoom on
    Block {
      BlockType Import
      Name "In1"
      Position [75, 93, 105, 107]
      BackgroundColor "cyan"
      Port "1"
      PortWidth "+1"
      SampleTime "+1"
      DataType "auto"
      SignalType "auto"
      Interpolate on
    }
    Block {
      BlockType Constant
      Name "Constant"
      Position [80, 134, 110, 156]
      BackgroundColor "red"
      Value "160"
    }
    Block {
      BlockType DataStoreMemory
      Name "DataStore\nMemory"
      Position [430, 35, 462, 65]
      BackgroundColor "orange"
      DataStoreName "AGC5"
      InitialValue "0"
    }
    Block {
      BlockType DataStoreRead
      Name "DataStore\nRead"
      Position [210, 155, 240, 185]
      BackgroundColor "orange"
      DataStoreName "AGC5"
      SampleTime "1/Fs"
    }
    Block {
      BlockType DataStoreWrite
      Name "DataStore\nWrite"
      Position [435, 175, 465, 205]
      BackgroundColor "orange"
      DataStoreName "AGC5"
      SampleTime "1/Fs"
    }
    Block {
      BlockType Reference
      Name "Integer Delay"
      Ports [1, 1, 0, 0, 0]
      Position [200, 67, 245, 103]
      BackgroundColor "red"
      SourceBlock "dspbdspl2/Integer Delay"
     SourceType "Integer"
      Delay 
        delay "160"
        ic "0"
        frame off
        df on
        numChans "1"
      }
      Block {
        BlockType Product
        Name "Product"
        Ports [2, 1, 0, 0, 0]
        Position [145, 92, 175, 123]
        BackgroundColor "green"
        Inputs "*"
        SaturateOnIntegerOverflow on
      }
    }
  }
}

212
Data Type "auto"
Signal Type "auto"
Interpolate on

Block {
    Block Type Constant
    Name "Constant"
    Position [86, 134, 110, 156]
    Background Color "red"
    Value "3520"
}
Block {
    Block Type DataStoreMemory
    Name "Data"
    Store Name "Memory"
    Position [430, 35, 462, 65]
    Background Color "orange"
    Data Store Name "AGCS5"
    Initial Value "0"
}
Block {
    Block Type DataStoreRead
    Name "Data"
    Store Name "Read"
    Position [210, 155, 240, 185]
    Background Color "orange"
    Data Store Name "AGCS5"
    Sample Time "1/Fs"
}
Block {
    Block Type DataStoreWrite
    Name "Data"
    Store Name "Write"
    Position [435, 175, 465, 205]
    Background Color "orange"
    Data Store Name "AGCS5"
    Sample Time "1/Fs"
}
Block {
    Block Type Reference
    Name "Integer Delay"
    Ports [0, 1, 0, 0, 0]
    Position [200, 67, 245, 103]
    Background Color "red"
    Source Block "dspbdspl/Integer Delay"
    Source Type "Integer"
    Delay Delay "3520"
    ic "0"
    frame off
    df on
    num Chans "1"
}
Block {
    Block Type Product
    Name "Product"
    Ports [2, 1, 0, 0, 0]
    Position [145, 92, 175, 123]
    Background Color "green"
    Inputs "*"
Open
ModelBrowserVisibility off
ModelBrowserWidth 200
ScreenColor "lightBlue"
PaperOrientation "landscape"
PaperPositionMode "auto"
PaperType "usletter"
PaperUnits "inches"
ZoomFactor "100"
AutoZoom on

Block
  BlockType Import
  Name "In1"
  Position [25, 68, 55, 82]
  BackgroundColor "cyan"
  Port "1"
  PortWidth "1"
  SampleTime "-1"
  DataType "auto"
  SignalType "auto"
  Interpolate on

Block
  BlockType Constant
  Name "Constant1"
  Position [100, 80, 120, 100]
  BackgroundColor "darkGreen"
  Value "0"

Block
  BlockType Fcn
  Name "Fcn"
  Position [185, 165, 245, 195]
  BackgroundColor "yellow"
  Expr "20*log10(u/(3.1623*power(10,-6)))"

Block
  BlockType RelationalOperator
  Name "Relational\nOperator"
  Position [145, 67, 175, 98]
  BackgroundColor "yellow"
  Operator "<="

Block
  BlockType Sum
  Name "Sum"
  Ports [2, 1, 0, 0, 0]
  Position [120, 162, 150, 193]
  BackgroundColor "green"
  IconShape "rectangular"
  Inputs ++
  SaturateOnIntegerOverflow on

Block
  BlockType Switch
  Name "Switch"
  Position [280, 70, 310, 100]
  BackgroundColor "orange"
  Threshold "1"

Block
  BlockType Output
  Name "Out1"
  Position [360, 78, 390, 92]
  BackgroundColor "cyan"
  Port "1"
  OutputWhenDisabled "held"
  InitialOutput "[]"

Line
  SrcBlock "Fcn"
  SrcPort 1
  Points [15, 0]
  DstBlock "Switch"
  DstPort 3

Line
  SrcBlock "In1"
  SrcPort 1
  Points [10, 0]
  Branch
  Points [0, -45; 195, 0]
  DstBlock "Switch"
  DstPort 1

Branch
  DstBlock "Relational\nOperator"
  DstPort 1
  Brach
  Points [0, 110]
  DstBlock "Sum"
  DstPort 2

Line
  SrcBlock "Constant1"
  SrcPort 1
  DstBlock "Relational\nOperator"
  DstPort 2

Line
  SrcBlock "Relational\nOperator"
  SrcPort 1
  Points [30, 0]
  Branch
  Points [0, 50; -105, 0]
  DstBlock "Sum"
  DstPort 1

Line
  SrcBlock "Sum"
  SrcPort 1
  DstBlock "Fcn"
  DstPort 1

Line
  SrcBlock "Switch"
  SrcPort 1
  DstBlock "Out1"
null
ZoomFactor "100"
AutoZoom on

Block {
  BlockType Import
  Name "In1"
  Position [150, 78, 180, 92]
  BackgroundColor "cyan"
  Port "1"
  PortWidth "-1"
  SampleTime "-1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}

Block {
  BlockType Import
  Name "In2"
  Position [710, 298, 740, 312]
  BackgroundColor "cyan"
  Port "2"
  PortWidth "-1"
  SampleTime "-1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}

Block {
  BlockType Import
  Name "In3"
  Position [910, 298, 835, 312]
  BackgroundColor "cyan"
  Port "3"
  PortWidth "-1"
  SampleTime "-1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}

Block {
  BlockType Import
  Name "In4"
  Position [805, 298, 835, 312]
  BackgroundColor "cyan"
  Port "4"
  PortWidth "-1"
  SampleTime "-1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}

Block {
  BlockType Abs
  Name "Abs"
  Position [215, 70, 245, 100]
  BackgroundColor "red"
}

Block {
  BlockType Constant
  Name "Constant"
  Position [710, 195, 740, 225]
  BackgroundColor "green"
  Value "1"
}

Block {
  BlockType Constant
  Name "Constant1"
  Position [910, 350, 940, 380]
  BackgroundColor "green"
  Value "330"
}

Block {
  BlockType SubSystem
  Name "FastMoving"
  Ports [1, 1, 0, 0, 0]
  Position [270, 57, 300, 113]
  BackgroundColor "orange"
  DropShadow on
  ShowPortLabels off
  System {
    Name "FastMoving"
    Location [75, 131, 863, 472]
    Open off
    ModelBrowserVisibility off
    ModelBrowserWidth 200
    ScreenColor "lightBlue"
    PaperOrientation "landscape"
    PaperPositionMode "auto"
    PaperType "usletter"
    PaperUnits "inches"
    ZoomFactor "100"
    AutoZoom on
    Block {
      BlockType Import
      Name "In1"
      Position [75, 93, 105, 107]
      BackgroundColor "cyan"
      Port "1"
      PortWidth "-1"
      SampleTime "-1"
      DataType "auto"
      SignalType "auto"
      Interpolate on
    }
    Block {
      BlockType Constant
      Name "Constant"
      Position [80, 134, 110, 156]
      BackgroundColor "red"
      Value "160"
    }
    Block {
      BlockType DataStoreMemory
      Name "DataStore\nMemory"
      Store\nMemory "Data"
      Store\nRead "Data"
      Store\nWrite "Data"
      DataStoreName "AGC6"
      InitialValue "0"
      Position [430, 35, 462, 65]
      BackgroundColor "orange"
      DataStoreName "AGC6"
      SampleTime "1/fs"
    }
  }
}

Block {
  BlockType DataStoreRead
  Name "DataStore\nRead"
  Position [210, 155, 240, 185]
  BackgroundColor "orange"
  DataStoreName "AGC6"
  SampleTime "1/fs"
}

Block {
  BlockType DataStoreWrite
  Name "DataStore\nWrite"
  Position [210, 155, 240, 185]
  BackgroundColor "orange"
  DataStoreName "AGC6"
  SampleTime "1/fs"
}

222
Block {
  BlockType Product
  Name "Product1"
  Ports [2, 1, 0, 0, 0]
  Position [915, 72, 945, 103]
  BackgroundColor "yellow"
  Inputs "*"/.
  SaturateOnIntegerOverflow on
}

Block {
  BlockType Product
  Name "Product2"
  Ports [2, 1, 0, 0, 0]
  Position [995, 252, 1025, 283]
  BackgroundColor "yellow"
  Inputs "*"/.
  SaturateOnIntegerOverflow on
}

Block {
  BlockType Product
  Name "Product3"
  Ports [2, 1, 0, 0, 0]
  Position [615, 67, 645, 98]
  BackgroundColor "green"
  Inputs "2"
  SaturateOnIntegerOverflow on
}

Block {
  BlockType Product
  Name "Product4"
  Ports [2, 1, 0, 0, 0]
  Position [1020, 322, 1050, 353]
  BackgroundColor "yellow"
  Inputs "*"/.
  SaturateOnIntegerOverflow on
}

Block {
  BlockType RelationalOperator
  Name "Relational\nOperator3"
  Position [510, 142, 540, 173]
  BackgroundColor "yellow"

OutputWhenDisabled "held"
InitialOutput "0"

Name "Sum"
Ports [2, 1, 0, 0, 0]
Position [935, 258, 955, 287]
BackgroundColor "orange"
ShowName off
IconShape "rectangular"
Inputs "++"
SaturateOnIntegerOverflow on

Block {
BlockType Sum
Name "Sum1"
Ports [2, 1, 0, 0, 0]
Position [1055, 203, 1075, 232]
BackgroundColor "orange"
ShowName off
IconShape "rectangular"
Inputs "++"
SaturateOnIntegerOverflow on

Block {
BlockType Sum
Name "Sum2"
Ports [2, 1, 0, 0, 0]
Position [1130, 213, 1150, 242]
BackgroundColor "orange"
ShowName off
IconShape "rectangular"
Inputs "++"
SaturateOnIntegerOverflow on

Block {
BlockType Sum
Name "Sum3"
Ports [2, 1, 0, 0, 0]
Position [1075, 293, 1095, 322]
BackgroundColor "orange"
ShowName off

}
IconShape "rectangular"
Inputs "**"
SaturateOnIntegerOverflow on
}
Block {
BlockType Sum
Name "Sum8"
Ports [2, 1, 0, 0, 0]
Position [675, 107, 705, 138]
BackgroundColor "green"
IconShape "rectangular"
Inputs "**"
SaturateOnIntegerOverflow on
}
Block {
BlockType Switch
Name "Switch"
Position [1000, 75, 1030, 105]
BackgroundColor "yellow"
NamePlacement "alternate"
ShowName off
Threshold "10.**8"
}
Block {
BlockType SubSystem
Name "DB Conv"
Ports [1, 1, 0, 0, 0]
Position [125, 63, 350, 97]
BackgroundColor "darkGreen"
DropShadow on
ShowPortLabels off
System {
Name "DB Conv"
Location [8, 74, 474, 399]
Open off
ModelBrowserVisibility off
ModelBrowserWidth 200
ScreenColor "lightBlue"
PaperOrientation "landscape"
PaperPositionMode "auto"
PaperType "usletter"
PaperUnits "inches"
ZoomFactor "100"
AutoZoom on
Block {
BlockType Import
Name "In1"
Position [25, 68, 55, 82]
BackgroundColor "cyan"
Port "1"
PortWidth "-1"
SampleTime "-1"
DataType "auto"
SignalType "auto"
Interpolate on
}
Block {
BlockType Constant
Name "Constant1"
Position [100, 80, 120, 100]
BackgroundColor "darkGreen"
Value "0"
}
Block {
BlockType Fcn
Name "Fcn"
Position [185, 165, 245, 195]
BackgroundColor "yellow"
Expr "20*log10(w/(3.1623*power(10,-6)))"
}
Block {
BlockType RelationalOperator
Name "Relational\nOperator"
Position [145, 67, 175, 98]
BackgroundColor "yellow"
Operator "<="
}
Block {
BlockType Sum
Name "Sum"
Ports [2, 1, 0, 0, 0]
Position [120, 162, 150, 193]
BackgroundColor "green"
IconShape "rectangular"
Inputs "**"
SaturateOnIntegerOverflow on
}
Block {
BlockType Switch
Name "Switch"
Position [280, 70, 310, 100]
BackgroundColor "orange"
Threshold "1"
}
Block {
BlockType Outport
Name "Out1"
Position [360, 78, 390, 92]
BackgroundColor "cyan"
Port "1"
OutputWhenDisabled "held"
InitialOutput "[]"
}
Line {
SrcBlock "Fcn"
SrcPort 1
Points [15, 0]
DstBlock "Switch"
DstPort 3
}
Line {
SrcBlock "In1"
SrcPort 1
Points [10, 0]
Branch {
Points [0, -45, 195, 0]
DstBlock "Switch"
DstPort 1
}
Branch {
DstBlock "Relational\nOperator"
DstPort 1
}
Branch {
Points [0, 110]
Line 
SrcBlock  "In1"
SrcPort  1
Points  [10, 0]
Branch 
Points  [0, -45; 195, 0]
DstBlock  "Switch"
DstPort  1
Branch 
DstBlock  "Relational\nOperator"
DstPort  1
Branch 
Points  [0, 110]
DstBlock  "Sum"
DstPort  2

} 

Line 
SrcBlock  "Constant1"
SrcPort  1
DstBlock  "Relational\nOperator"
DstPort  2

} 

Line 
SrcBlock  "In5"
SrcPort  1
Points  [20, 0]
DstBlock  "Sum"
DstPort  2

} 

Line 
SrcBlock  "Abs"
SrcPort  1
Points  [5, 0]
DstBlock  "FastMoving"
DstPort  1

} 

Line 
SrcBlock  "Sum"
SrcPort  1
DstBlock  "Product1"
DstPort  2

} 

Line 
SrcBlock  "Switch"
SrcPort  1
DstBlock  "Out1"
DstPort  1

} 

Block 
BlockType  Constant
Name  "effect"
Position  [370, 210, 390, 230]
BackColor  "red"
Value  "6"

} 

Block 
BlockType  Outport
Name  "Out1"
Position  [1415, 88, 1445, 102]
BackColor  "cyan"
Port  "1"
OutputWhenDisabled  "held"
Points [10, 0]
Branch {
  DstBlock "Pcm1"
  DstPort 1
}
Branch {
  Points [0, 35; -40, 0; 0,
          170]
  DstBlock "Product4"
  DstPort 1
}
}
Block {
  BlockType SubSystem
  Name "homomorphic\n  BackgroundColor *orange*
  DropShadow on
  ShowPortLabels off
  System {
    Name "homomorphic\n    Location [107, 78, 883, 455]
    Open off
    ModelBrowserVisibility off
    ModelBrowserWidth 200
    ScreenColor "lightBlue"
    PaperOrientation "landscape"
    PaperPositionMode "auto"
    PaperType "usletter"
    PaperUnits "inches"
    ZoomFactor "100"
    AutoZoom on
  }
  Block {
    BlockType Import
    Name "In1"
    Position [150, 78, 180, 92]
    BackgroundColor "cyan"
    Port "1"
    PortWidth "1"
    SampleTime "1"
    DataType "auto"
    SignalType "auto"
    Interpolate on
  }
  Block {
    BlockType Import
    Name "In2"
    Position [270, 57, 300, 113]
    BackgroundColor "cyan"
    Port "2"
    PortWidth "1"
    SampleTime "1"
    DataType "auto"
    SignalType "auto"
    Interpolate on
  }
}
Block {
  BlockType Import
  Name "In3"
  Position [710, 298, 740, 312]
  BackgroundColor "cyan"
  Port "3"
  PortWidth "1"
  SampleTime "1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}
Block {
  BlockType Import
  Name "Out"
  Position [805, 298, 835, 312]
  BackgroundColor "cyan"
  Port "4"
  PortWidth "1"
  SampleTime "1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}
Block {
  BlockType Import
  Name "In4"
  Position [85, 93, 105, 107]
  BackgroundColor "cyan"
  Port "1"
  PortWidth "1"
  SampleTime "1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}
SignalType "auto"
Interpolate on
}
Block {
  BlockType Import
  Name "In5"
  Position [865, 298, 895, 312]
  BackgroundColor "cyan"
  Port "4"
  PortWidth "1"
  SampleTime "1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}
Block {
  BlockType Abs
  Name "Abs"
  Position [215, 70, 245, 100]
  BackgroundColor "red"
}
Block {
  BlockType Constant
  Name "Constant"
  Position [710, 195, 740, 225]
  BackgroundColor "green"
  Value "1"
}
Block {
  BlockType Constant
  Name "Constant1"
  Position [910, 350, 940, 380]
  BackgroundColor "green"
  Value "330"
}
Block {
  BlockType SubSystem
  Name "FastMoving"
  Ports [1, 1, 0, 0, 0]
  Position [270, 57, 300, 113]
  BackgroundColor "cyan"
  Port "1"
  PortWidth "1"
  SampleTime "1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}
Block {
  BlockType SubSystem
  Name "FastMoving"
  Location [77, 129, 865, 470]
  Open off
  ModelBrowserVisibility off
  ModelBrowserWidth 200
  ScreenColor "lightBlue"
  PaperOrientation "landscape"
  PaperPositionMode "auto"
  PaperType "usletter"
  PaperUnits "inches"
  ZoomFactor "100"
  AutoZoom on
}
Block {
  BlockType Import
  Name "In1"
  Position [75, 93, 105, 107]
  BackgroundColor "cyan"
  Port "1"
  PortWidth "1"
  SampleTime "1"
  DataType "auto"
  SignalType "auto"
  Interpolate on
}


Block {
  BlockType Constant
  Name "Constant"
  Position [80, 134, 110, 156]
  BackgroundColor "red"
  Value "160"
}
Block {
  BlockType DataStoreMemory
  Name "DataStore\nMemory"
  Position [430, 35, 462, 65]
  BackgroundColor "orange"
  DataStoreName "AGC7"
  InitialValue "0"
}
Block {
  BlockType DataStoreRead
  Name "DataStore\nRead"
  Position [210, 155, 240, 185]
  BackgroundColor "orange"
  DataStoreName "AGC7"
  SampleTime "1/Fs"
}
Block {
  BlockType DataStoreWrite
  Name "DataStore\nWrite"
  Position [435, 175, 465, 205]
  BackgroundColor "orange"
  DataStoreName "AGC7"
  SampleTime "1/Fs"
}
Block {
  BlockType Reference
  Name "Integer Delay"
  Position [200, 67, 245, 103]
  BackgroundColor "red"
  SourceBlock "dspbdspl/Integer Delay"
  SourceType "Integer"
  Delay delay "160"
  ic "0"
  frame off
  df on
  numChans "1"
}
Block {
  BlockType Product
  Name "Product"
  Position [145, 92, 175, 123]
  BackgroundColor "green"
  Inputs "*/"
  SaturateOnIntegerOverflow on
}
Block {
  BlockType Sum
  Name "Sum"
  Ports [2, 1, 0, 0, 0]
  Position [375, 121, 405, 154]
  BackgroundColor "green"
  DropShadow on
  IconShape "rectangular"
  Inputs "++"
  SaturateOnIntegerOverflow on
}
Block {
  BlockType Outport
  Name "Outl"
  Position [490, 133, 520, 147]
  BackgroundColor "cyan"
  Port "1"
  OutputWhenDisabled "held"
  InitialOutput "0"
}
Line {
  SrcBlock "Constant"
  SrcPort 1
  Points [15, 0]
  DstBlock "Product"
  DstPort 2
}
Line {
  SrcBlock "In1"
  SrcPort 1
  DstBlock "Product"
  DstPort 1
}
Line {
  SrcBlock "DataStore\nRead"
  SrcPort 1
  Points [20, 0]
  DstBlock "Suml"
  DstPort 2
}
Line {
  SrcBlock "Integer Delay"
  SrcPort 1
  DstBlock "Suml"
  DstPort 1
}
Line {
  SrcBlock "Suml"
  SrcPort 1
  Points [45, 0]
  DstBlock "Sum"
  DstPort 1
}

Block {
    BlockType    Product
    Name         "Product2"
    Ports        [2, 1, 0, 0, 0]
    Position     [1175, 82, 1205,
113]
    BackgroundColor  "yellow"
    Inputs        ---
    SaturateOnIntegerOverflow on
}

Block {
    BlockType    Product
    Name         "Product3"
    Ports        [2, 1, 0, 0, 0]
    Position     [1350, 77, 1380,
108]
    BackgroundColor  "yellow"
    Inputs        ---
    SaturateOnIntegerOverflow on
}

Block {
    BlockType    Product
    Name         "Product4"
    Ports        [2, 1, 0, 0, 0]
    Position     [1020, 322, 1050,
353]
    BackgroundColor  "yellow"
    Inputs        /*
    SaturateOnIntegerOverflow on
}

Block {
    BlockType    RelationalOperator
    Name         "Relational\nOperator3"
    Position     [510, 142, 540, 173]
    BackgroundColor  "yellow"
    Operator     ">="
}

Block {
    BlockType    SubSystem
    Name         "SlowMoving"
    Ports        [1, 1, 0, 0, 0]
    Position     [270, 137, 300, 193]
    BackgroundColor  "orange"
    DropShadow    on
    ShowPortLabels off
    System {
        Name        "SlowMoving"
        Location    [79, 127, 867,
468]
        Open        off
        ModelBrowserVisibility off
        ModelBrowserWidth  200
        ScreenColor   "lightBlue"
        PaperOrientation "landscape"
        PaperPositionMode "auto"
        PaperType    "usletter"
        PaperUnits   "inches"
        ZoomFactor   "100"
        AutoZoom     on
    Block {
        BlockType    Import
        Name         "In1"
        Position     [75, 93,
105, 107]
        BackgroundColor  "cyan"
        Port          "1"
        PortWidth     "-1"
        SampleTime    "-1"
    Block {
        BlockType    Constant
        Name         "Constant"
        Position     [80, 134,
110, 156]
        BackgroundColor  "red"
        Value         "3520"
    Block {
        BlockType    DataStoreMemory
        Name         "Data
Store\nMemory"
        Position     [430, 35,
462, 65]
        BackgroundColor  "orange"
        DataStoreName   "AGCS7"
        InitialValue    "0"
    Block {
        BlockType    DataStoreRead
        Name         "Data
Store\nRead"
        Position     [210, 155,
240, 185]
        BackgroundColor  "orange"
        DataStoreName   "AGCS7"
        SampleTime     "1/Fs"
    Block {
        BlockType    DataStoreWrite
        Name         "Data
Store\nWrite"
        Position     [435, 175,
465, 205]
        BackgroundColor  "orange"
        DataStoreName   "AGCS7"
        SampleTime     "1/Fs"
    Block {
        BlockType    Reference
        Name         "Integer Delay"
        Ports        [1, 1, 0, 0, 0]
        Position     [200, 67,
245, 103]
        BackgroundColor  "red"
        SourceBlock   "dspbdsp2/Integer Delay"
        SourceType    "Integer
Delay"
        delay        "3520"
        ic           "0"
        frame       off
        df           on
        numChans    "1"
    Block {
        BlockType    Product
        Name         "Product"
        Ports        [2, 1, 0, 0, 0]
        Position     [145, 92,
175, 123]
        BackgroundColor  "green"
        Inputs        "/*"
BlockType: SubSystem
Name: "FastMoving"
Ports: [1, 1, 0, 0, 0]
Position: [270, 57, 300, 113]
BackgroundColor: "orange"
DropShadow: on
ShowPortLabels: off
System:
  Name: "FastMoving"
  Location: [79, 127, 857, 468]
Open: off
ModelBrowserVisibility: off
ModelBrowserWidth: 200
ScreenColor: "lightBlue"
PaperOrientation: "landscape"
PaperPositionMode: "auto"
PaperType: "usletter"
PaperUnits: "inches"
ZoomFactor: "100"
AutoZoom: on
Block:
  BlockType: Import
  Name: "In1"
  Position: [75, 93, 105, 107]
BackgroundColor: "cyan"
Port: [1]
PortWidth: "-1"
SampleTime: "-1"
DataType: "auto"
SignalType: "auto"
Interpolate: on
Block:
  BlockType: Constant
  Name: "Constant"
  Position: [80, 134, 110, 156]
BackgroundColor: "red"
Value: "160"
Block:
  BlockType: DataStoreMemory
  Name: "DataStore\Memory"
  Position: [430, 35, 462, 65]
BackgroundColor: "orange"
DataStoreName: "AGCS"
InitialValue: "0"
Block:
  BlockType: DataStoreRead
  Name: "DataStore\Read"
  Position: [210, 155, 240, 185]
BackgroundColor: "orange"
DataStoreName: "AGCS"
SampleTime: "1/Fs"
Block:
  BlockType: DataStoreWrite
  Name: "DataStore\Write"
  Position: [435, 175, 465, 205]
BackgroundColor: "orange"
DataStoreName: "AGCS"
SampleTime: "1/Fs"
Block:
  BlockType: Reference
  Name: "Integer Delay"
  Ports: [1, 1, 0, 0, 0]
  Position: [200, 67, 245, 103]
BackgroundColor: "red"
SourceBlock: "dspbsdsp2/Integer Delay"
SourceType: "integer"
Delay:
  delay: "160"
  ic: "0"
  frame: off
  df: on
  numChans: "1"
Block:
  BlockType: Product
  Name: "Product"
  Ports: [2, 1, 0, 0, 0]
  Position: [145, 92, 175, 123]
BackgroundColor: "green"
Inputs: "/
SaturateOnIntegerOverflow: on
Block:
  BlockType: Sum
  Name: "Sum"
  Ports: [2, 1, 0, 0, 0]
  Position: [375, 121, 405, 154]
BackgroundColor: "green"
DropShadow: on
IconShape: "rectangular"
Inputs: "/
SaturateOnIntegerOverflow: on
Block:
  BlockType: Sum
  Name: "Sum1"
  Ports: [2, 1, 0, 0, 0]
  Position: [280, 77, 310, 108]
BackgroundColor: "green"
DropShadow: on
IconShape: "rectangular"
Inputs: "/
SaturateOnIntegerOverflow: on
Block:
  BlockType: Outport
  Name: "Out1"
  Position: [490, 133, 520, 147]
BackgroundColor: "cyan"
Port: [1]
OutputWhenDisabled: "held"
InitialOutput: "0"
Line: 

243
BlockType
DataStoreWrite
Name "Data"
Position [435, 175]
BackgroundColor "orange"
DataStoreName "AGCS8"
SampleTime "1/Fs"

465, 205
Block {
BlockType Reference
Name "Integer Delay"
Ports [1, 1, 0, 0, 0]
Position [200, 67]
BackgroundColor "red"
SourceBlock "dsdpdsp2/Integer Delay"
SourceType "Integer"
Delay
delay "3520"
ic "0"
frame off
df on
numChans "1"
}

245, 103
Block {
BlockType Product
Name "Product"
Ports [2, 1, 0, 0, 0]
Position [145, 92]
BackgroundColor "green"
Inputs "*/*" SaturateOnIntegerOverflow on
}

175, 123
Block {
BlockType Sum
Name "Sum"
Ports [2, 1, 0, 0, 0]
Position [375, 121]
BackgroundColor "green"
DropShadow on
IconShape "rectangular"
Inputs "++" SaturateOnIntegerOverflow on
}

405, 154
Block {
BlockType Sum
Name "Sum"
Ports [2, 1, 0, 0, 0]
Position [280, 77]
BackgroundColor "green"
DropShadow on
IconShape "rectangular"
Inputs "--" SaturateOnIntegerOverflow on
}

310, 108
Block {
BlockType Sum
Name "Sum"
Ports [2, 1, 0, 0, 0]
Position [380, 133]
BackgroundColor "cyan"
Port "1"
}

OutputWhenDisabled "held"
InitialOutput "0"

Line {
SrcBlock "Constant"
SrcPort 1
Points [15, 0]
DstBlock "Product"
DstPort 2
}

Line {
SrcBlock "In1"
SrcPort 1
DstBlock "Product"
DstPort 1
}

Line {
SrcBlock "Data"
Store\nRead
SrcPort 1
Points [20, 0]
DstBlock "Sum1"
DstPort 2
}

Line {
SrcBlock "Integer"
Delay
SrcPort 1
DstBlock "Sum1"
DstPort 1
}

Line {
SrcBlock "Sum1"
SrcPort 1
Points [45, 0]
DstBlock "Sum"
DstPort 1
}

Line {
SrcBlock "Product"
SrcPort 1
Points [5, 0]
DstBlock "Integer Delay"
DstPort 1
}

Branch {
Points [0, 35]
DstBlock "Sum"
DstPort 2
}

Branch {
Points [10, 0]
DstBlock "Data"
Store\nWrite
DstPort 1
}

Branch {
DstBlock "Out1"
DstPort 1
}

Block {
BlockType Sum
}
<table>
<thead>
<tr>
<th>Name</th>
<th>&quot;Sum&quot;</th>
<th>IconShape</th>
<th>&quot;rectangular&quot;</th>
<th>Inputs</th>
<th>&quot;++&quot;</th>
<th>SaturateOnIntegerOverflow on</th>
</tr>
</thead>
<tbody>
<tr>
<td>Blocks</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Block</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BlockType</td>
<td>Sum</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Name</td>
<td>&quot;Sum1&quot;</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ports</td>
<td>[2, 1, 0, 0, 0]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Position</td>
<td>[935, 258, 955, 287]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Block</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BlockType</td>
<td>Sum</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Name</td>
<td>&quot;Sum1&quot;</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ports</td>
<td>[2, 1, 0, 0, 0]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Position</td>
<td>[1055, 203, 1075]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Block</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BlockType</td>
<td>Sum</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Name</td>
<td>&quot;Sum2&quot;</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ports</td>
<td>[2, 1, 0, 0, 0]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Position</td>
<td>[2322, 582, 382, 233]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Block</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BlockType</td>
<td>Sum</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Name</td>
<td>&quot;Sum3&quot;</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ports</td>
<td>[2, 1, 0, 0, 0]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Position</td>
<td>[3222, 1095, 233]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
PortWidth "-1"
SampleTime "-1"
DataType "auto"
SignalType "auto"
Interpolate on

Block {
    BlockType Constant
    Name "Constant1"
    Position [100, 80, 120, 100]
    BackgroundColor "darkGreen"
    Value "0"
}

Block {
    BlockType Fcn
    Name "Fcnc"
    Position [185, 165, 245, 195]
    BackgroundColor "yellow"
    Expr "20*log10(u/(3.1623^power(10, -6)))"
}

Block {
    BlockType RelationalOperator
    Name "Relational\nOperator"
    Position [145.67, 175, 98]
    BackgroundColor "yellow"
    Operator "<="
}

Block {
    BlockType Sum
    Name "Sum"
    Ports [2, 1, 0, 0, 0]
    Position [120, 162, 150, 193]
    BackgroundColor "green"
    IconShape "rectangular"
    Inputs "--"
    SaturateOnIntegerOverflow on
}

Block {
    BlockType Switch
    Name "Switch"
    Position [280, 70, 310, 100]
    BackgroundColor "orange"
    Threshold "1"
}

Block {
    BlockType Outport
    Name "Out1"
    Position [360, 78, 390, 92]
    BackgroundColor "cyan"
    Port "1"
    OutputWhenDisabled "held"
    InitialOutput "[1]"
}

Line {
    SrcBlock "In1"
    SrcPort 1
    Points [10, 0]
    Branch {
        Points [0, -45; 195, 0]
        DstBlock "Switch"
        DstPort 1
    }
    DstBlock "Relational\nOperator"
    DstPort 1
    Branch {
        Points [0, 110]
        DstBlock "Sum"
        DstPort 2
    }
}

Line {
    SrcBlock "Constant1"
    SrcPort 1
    DstBlock "Relational\nOperator"
    DstPort 2
}

Line {
    SrcBlock "Relational\nOperator"
    SrcPort 1
    Points [30, 0]
    Branch {
        Points [0, 50; -105, 0]
        DstBlock "Sum"
        DstPort 1
    }
}

Line {
    SrcBlock "Sum"
    SrcPort 1
    DstBlock "Fcnc"
    DstPort 1
}

Line {
    SrcBlock "Switch"
    SrcPort 1
    DstBlock "Out1"
    DstPort 1
}

Line {
    SrcBlock "Fcnc"
    SrcPort 1
    Points [15, 0]
    DstBlock "Switch"
    DstPort 3
}

Block {
    BlockType Constant
    Name "effect"
    Position [370, 210, 390, 230]
    BackgroundColor "red"
    Value "6"
}

Block {
    BlockType Outport
    Name "Out1"
    Position [1415, 88, 1445, 102]
    BackgroundColor "cyan"
    Port "1"
    OutputWhenDisabled "held"
}
InitialOutput  "0"
}

Branch {
  Points  [0, -90]
  DstBlock  "Switch"
  DstPort  3
}

SourceBlock  "In1"

SourcePort  1

Points  [5, 0]

Branch {
  DstBlock  "Abs"
  DstPort  1
}

Branch {
  Points  [0, -80; 1145, 0]
  DstBlock  "Product3"
  DstPort  1
}

SourceBlock  "Sum2"

SourcePort  1

Points  [5, 0]

DstBlock  "Product2"

DstPort  2

SourceBlock  "Math\nFunction"

SourcePort  1

DstBlock  "Product2"

DstPort  1

SourceBlock  "Math\nFunction1"

SourcePort  1

DstBlock  "Product1"

DstPort  1

SourceBlock  "Math\nFunction1"

SourcePort  1

DstBlock  "Product3"

DstPort  2

SourceBlock  "Product"

SourcePort  1

Points  [25, 0]

Branch {
  DstBlock  "Switch"
  DstPort  2
}

Branch {
  Points  [0, -10]
  DstBlock  "Switch"
  DstPort  1
}

SourceBlock  "Switch"

SourcePort  1

DstBlock  "Math\nFunction"

DstPort  1

SourceBlock  "In4"

SourcePort  1

Points  [0, -40]

Branch {
  DstBlock  "Sum"
  DstPort  1
}

Branch {
  Points  [-20, 0]
  DstBlock  "Fcn"
  DstPort  1
}
Branch {
    Points [5, 0]
    Branch {
        Points [-10, 0; 0, 135]
            Branch {
                DstBlock "Sum12"
                DstPort 1
            }
            Branch {
                DstBlock "Product12"
                DstPort 2
            }
    }
    Branch {
        Points [50, 0]
        DstBlock "Relational\nOperator"
        DstPort 1
    }
}

Line {
    SrcBlock "dB Conv.-LF"
    SrcPort 1
    Points [0, 5; 35, 0; 0, 15; 100, 0]
    Branch {
        Points [0, 0]
        DstBlock "Relational\nOperator"
        DstPort 2
    }
    Branch {
        Points [0, 65]
        DstBlock "Sum12"
        DstPort 2
    }
    Branch {
        DstBlock "Product11"
        DstPort 1
    }
}

Line {
    SrcBlock "FastMoving"
    SrcPort 1
    Points [5, 0]
    DstBlock "dB Conv"
    DstPort 1
}

Line {
    SrcBlock "SlowMoving"
    SrcPort 1
    Points [5, 0]
    DstBlock "dB Conv.-LF"
    DstPort 1
}

Line {
    SrcBlock "Fcml"
    SrcPort 1
    Points [25, 0; 0, -45]
    DstBlock "Product"
    DstPort 1
}

Line {
    SrcBlock "Sum8"
    SrcPort 1
    DstBlock "Fcml"
    DstPort 1
    Points [10, 0]
    Branch {
        DstBlock "Product4"
        DstPort 1
    }
}

Line {
    SrcBlock "Subsystem"
    SrcPort 1
    DstBlock "homomorphic\nmultiplicative AGC1"
    DstPort 1
}

Line {
    SrcBlock "NHT1"
    SrcPort 1
    Points [50, 0]
    DstBlock "homomorphic\nmultiplicative AGC1"
    DstPort 4
}

Line {
    SrcBlock "UCL1"
    SrcPort 1
    Points [0, 10]
    DstBlock "homomorphic\nmultiplicative AGC1"
    DstPort 3
}

Line {
    SrcBlock "HL1"
    SrcPort 1
    Points [10, 0; 0, .25]
    DstBlock "homomorphic\nmultiplicative AGC1"
    DstPort 2
}

Line {
    SrcBlock "Subsystem"
    SrcPort 2
    Points [0, -5]
    DstBlock "homomorphic\nmultiplicative AGC2"
    DstPort 1
}

Line {
    SrcBlock "NHT2"
    SrcPort 1
    DstBlock "homomorphic\nmultiplicative AGC2"
    DstPort 4
}

Line {
    SrcBlock "HL2"
    SrcPort 1
    Points [10, 0; 0, -30]
    DstBlock "homomorphic\nmultiplicative AGC2"
    DstPort 2
}

Line {
    SrcBlock "UCL2"
    SrcPort 1
```
DstBlock
  "homomorphic\nmultiplicative AGC7"
  DstPort  2
}
Line {
  SrcBlock  "NHT7"
  SrcPort  1
  DstBlock
  "homomorphic\nmultiplicative AGC7"
  DstPort  4
}
Line {
  SrcBlock  "UCL7"
  SrcPort  1
  Points  [5, 0; 0, -15]
  DstBlock
  "homomorphic\nmultiplicative AGC7"
  DstPort  3
}
Line {
  SrcBlock  "Subsystem"
  SrcPort  8
  DstBlock
  "homomorphic\nmultiplicative AGC8"
  DstPort  1
}
Line {
  SrcBlock  "HL8"
  SrcPort  1
  Points  [0, -30]
  DstBlock
  "homomorphic\nmultiplicative AGC8"
  DstPort  2
}
Line {
  SrcBlock  "NHT8"
  SrcPort  1
  DstBlock
  "homomorphic\nmultiplicative AGC8"
  DstPort  4
}
Line {
  SrcBlock  "UCL8"
  SrcPort  1
  Points  [5, 0; 0, -15]
  DstBlock
  "homomorphic\nmultiplicative AGC8"
  DstPort  3
}
Line {
  SrcBlock
  "homomorphic\nmultiplicative AGC1"
  SrcPort  1
  Points  [160, 0; 0, 275]
  DstBlock
  "Sum"
  DstPort  1
}
Line {
  SrcBlock
  "homomorphic\nmultiplicative AGC2"
  SrcPort  1
  Points  [150, 0; 0, 210]
  DstBlock
  "Sum"
  DstPort  2
}
Line {
  SrcBlock
  "homomorphic\nmultiplicative AGC3"
  SrcPort  1
  Points  [140, 0; 0, 140]
  DstBlock
  "Sum"
  DstPort  3
}
Line {
  SrcBlock
  "homomorphic\nmultiplicative AGC4"
  SrcPort  1
  Points  [125, 0; 0, 70]
  DstBlock
  "Sum"
  DstPort  4
}
Line {
  SrcBlock
  "homomorphic\nmultiplicative AGC6"
  SrcPort  1
  Points  [140, 0; 0, -70]
  DstBlock
  "Sum"
  DstPort  6
}
Line {
  SrcBlock
  "homomorphic\nmultiplicative AGC7"
  SrcPort  1
  Points  [155, 0; 0, -140]
  DstBlock
  "Sum"
  DstPort  7
}
Line {
  SrcBlock
  "homomorphic\nmultiplicative AGC8"
  SrcPort  1
  Points  [180, 0]
  DstBlock
  "Sum"
  DstPort  8
}
Line {
  SrcBlock  "Sum"
  SrcPort  1
  DstBlock  "Output signal"
  DstPort  1
}
Line {
  SrcBlock
  "homomorphic\nmultiplicative AGC9"
  SrcPort  1
  DstBlock
  "Sum"
  DstPort  5
}
Line {
  SrcBlock
  "Sound"
  SrcPort  1
  DstBlock  "Subsystem"
  DstPort  1
}
Annotation {
  Position  [736, 56]
  ForegroundColor  "white"
  BackgroundColor  "red"
  Text  "HOMOMORPHIC\nMULTIPlicative\nAGC\nHEARING"
  "SIMULATOR"
  FontSize  14
  FontWeight  "bold"
  FontAngle  "italic"
}
254
```
NAME: Erkan Onat

PLACE OF BIRTH: Izmir, Turkey

YEAR OF BIRTH: 1975

EDUCATION:

Istanbul Anadolu High School, Istanbul, Turkey 1986-1993

Istanbul Technical University, Istanbul, Turkey 1993-1995 Transferred to Bogazici University

Bogazici University, Istanbul, Turkey 1995-1998 B.Sc. in Electrical Engineering

University of Windsor, Windsor, Ontario 1998-2001 M.A.Sc. in Electrical Engineering