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Dynamic Range Compression FRM Filter for Hearing Aid with Multiband Adaptability

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Abstract: Hearing aid technology that helps people has become more important lately. Many people struggle with hearing sounds. This tells about a new way to make quiet and loud sounds easier to hear with changes over different tones. The plan uses adjustable sections to help with all the different sounds around us that are always changing. The FRM filter separates sound into parts based on their pitches. It is the main part that changes how loud and soft the sounds are. Unlike compressors that treat all sounds the same, this system splits sound into groups of pitches. Then it adjusts each group to make things easier to hear. This multiband change makes it possible to shape sound in a detailed way matching how someone hears. This gives a truer picture when sounds blend together making things complex. The hearing aid learns from the user over time. It pays close attention to what is easy or hard for the person to hear. Then it makes small changes to make sounds clearer. This helps the user understand speech better and enjoy the sound more. No two people hear exactly the same, so the aid adjusts to fit each individual. The proposed design incorporates the most advanced sound methods to lessen issues and mistakes, helping the quality of the adjusted audio. We checked the system utilizing recreations and tests in the real world, and it showed generally better outcomes in diverse listening circumstances. This new sound filter for hearing aids uses different techniques to help people with different types of hearing loss. It changes how loud or soft sounds are based on the person's needs. Special algorithms also adjust the filter over time based on how the person reacts to sounds. These updates help sounds be clearer and more comfortable. The filter combines advanced sound processing with adaptive changes. This system should help hearing aids work better for each individual. It has potential to improve how hearing aids help people live with hearing problems.

Keywords*: *Hearing aids, (MDRC), RFRMF, Audiogram-based customization, MATLAB GUI for auditory compensation

1. Introduction

The escalating prevalence of hearing loss, affecting 5% of the global population, necessitates effective rehabilitation strategies. Defined as impairment exceeding 35 decibels (dB), disabling hearing loss underscores the urgent requirement for enhanced auditory assistive devices. Traditional hearing aids, characterized by static functionality and limited personalization, have often left users dissatisfied due to their inability to adapt to diverse environments and individual hearing profiles. Addressing the challenges posed by inner ear damage, specifically to hair cells, hearing aids have emerged as a standard solution for individuals with hearing loss. Configurability has become imperative to cater to diverse hearing problems using the same device [1]. The filter bank within hearing aids should offer sharp transitions with minimal complexity and filter order, making Frequency Response Masking (FRM) a fitting technique to meet these requirements.

The realm of auditory assistive technology has changed greatly with the creation of a new kind of Dynamic Range Compression (DRC) Frequency Response Masking (FRM) Filter for hearing aids. This filter can adjust to different sound bands. As hearing loss affects more people worldwide, over 5% of the population, the need for help with hearing has become very important. Hearing loss that is more than 35 decibels (dB) makes listening very hard. This shows why new and customized devices are needed. Old hearing aids were often said to be too fixed and not able to change enough. This caused problems for people using them in different places and with different hearing. The starting of Dynamic Range Compression (DRC) was a big step forward in improving hearing aid technology.

The filter bank [2]can sharply separate different sounds with few filters and less complexity, making it a good choice for hearing aids. The filter bank can easily change to suit different places and hearing tests. This helps hearing aids adjust exactly to where people are and what they can hear. Since some people with hearing loss have a hard time hearing quiet sounds but can still hear loud noises clearly, the filter bank separates sounds into bands. This helps give each person a detailed and customized listening experience. Software called a

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Graphical User Interface was created using MATLAB too. It gives a simple way to see and control settings on a computer. The GUI lets people bring in audio files, see how sounds are filtered, and adjust options based on hearing tests. It also shows how the filter bank changes the volume in each band. And users can finely tune the volume points for each band to fix mistakes and have more control over their hearing aid settings.

2. Related Work

The merging of a filter that changes loudness levels and allows splitting into frequency bands addresses difficulties faced by those with hearing difficulties. This imaginative method boosts sounds and enhances understanding besides supplying an extremely customizable answer by including masking of freque-ncies [3]. Combining loudness change and frequency masking, along with an easy-to-use interface, displays not just an engineering improvement but in addition a thoughtful reaction to the various and progressing needs of people with hearing loss, guaranteeing a more inclusive and personalized experience of what they can hear.

Table 1: Summary of Related Work

Method	Approach	Finding	Scope
DRC Integration [4]	Incorporation of DRC in filter design	Enhanced speech comprehension and sound quality	Adaptability to diverse hearing profiles
FRM Implementation [5]	Integration of FRM in filter bank design	Reconfigurable and adaptive framework for multiband use	Tailored solutions for varied environments
Filter Bank Design [6]	Meticulous arrangement of sub-bands	Optimization of matching performance	Customization based on individual audiograms
GUI Development [7]	MATLAB-based graphical interface	User-friendly platform for visualization and parameter control	Enhanced user control and adjustment
Audiogram-based Customization [8]	Adjusting gain based on individual audiograms	Minimization of errors in gain adjustments	Improved user experience and satisfaction
Frequency Response Observation [9]	GUI facilitates frequency response analysis	Visualization of audio file responses based on audiograms	Real-time observation for user adjustments
Automatic Gain Adjustment [10]	Filter bank dynamically adjusts gain	Improved adaptability to changing auditory environments	Consistent performance in varying settings
Multiband FRM Filter Design [11]	FRM implementation across multiple bands	Precision in amplification and compression across frequencies	Customized solutions for varying frequency-dependent impairments
MATLAB-based Development [12]	Utilization of MATLAB for GUI creation	Efficient platform for visualization and parameter adjustment	Seamless integration with existing tools
User Satisfaction Evaluation [13]	Gathering user feedback on the system	Positive response to personalized and adaptable hearing aid	Insight for further refinements and advancements

3. Methodology

This paper presents a new way to improve how well hearing aids work. It creates a sound filter using Multiband Dynamic Range Compression. The filter changes the loudness of different sounds. The steps to make this new filter are explained below.

1. Implementing FRM Approach for Filterbank Design

First, we plan how to split the sound into parts using filters. This lets us separate the audio into different frequency ranges [15]. Some filters will pass higher pitches while blocking lower ones. Others will do the opposite. Together the filters will take the full sound and break it into pieces for closer examination.

2. Unraveling Signals with FRM Filterbank: A Breakdown Approach

At first, the sound signal goes through several steps to get it ready for processing. It goes to the FRM filterbank first. This special filterbank takes apart the sound into different subbands, where each one stands for a certain group of pitches. Breaking it down this way is important because then each pitch part in the signal can be singled out alone, letting later stages handle it better.

3. Adapting Performance through Predefined Value-Based Adjustments

The audio was modified for different frequency bands, using preset levels from testing many with hearing loss. The levels came from 105 people who had trouble hearing and how their ears handled different sounds.

4. Harnessing the Power of Multiband Dynamic Range Compression (MDRC)

Next, every single frequency band transmission that is created from the Filtered Reference Model (FRM) filterbank experiences a complex technique called Multiband Dynamic Range Compression (MDRC) [14]. This strategy adaptively changes the volume or addition of each frequency band transmission subject to its particular level. The main goal of MDRC is making sure that low volume noises are amplified to a bigger extent compared to louder noises. For powerful execution of MDRC, many pivotal factors need thought, such as Compression proportion, Attack time, Release time, and Knee point.

5. Subband Signal Recombination and Synthesis:

The signals separated into frequency ranges then went through additional processing to change their volumes. This brought them back together using a tool that mirrors the original filter system. It aims to make a better sound file customized for the user. By putting the frequency ranges back as a whole, this tool is key to reaching this goal.

This step-by-step process [16] shows a complete way to make hearing aids work better. It uses different parts to meet different needs. First, it separates sounds into groups. Then, it adjusts the volume for each person based on their hearing test. It also uses compression to make quiet and loud sounds better. Putting it all together gives a sound that is customized for each user. This improved hearing aid design can really help people who have trouble hearing.

A. FRM Based Filter Bank Design:

This hearing aid design's notable quality involves the careful division of frequencies within the filter bank. Engineers considered each user's hearing test results. The goal was to best match performance for people with different hearing problems. Tailoring the filter's responses fits an individual's hearing test aligns extra bands in the low, middle, and high ranges. This shows flexibility. Being able to change more is very useful for fixing age-related high hearing problems, as extra high bands can help matching work better. The filter bank deals with hearing issues across many frequencies by smartly placing subbands based on personal hearing tests. This personal way helps optimize matching, especially for addressing lost high hearing in older people. Adding extra high bands displays the filter bank's ease of use and effectiveness. This marks big progress in improving how well hearing aids can help [17].

The new design highlighted both the complex details of how the filter bank works and why it's important to create options that meet different people's needs who have trouble hearing. The suggested change would improve hearing aids a lot by changing how the filter bank functions to match each person's specific hearing, to make sure they get better and personalized ways to experience sounds.

Table 2: Summary of Sixteen Sub band Cut off Frequency

Band	Frequency Lower 3 dB	Frequency Upper 3 dB
1	-260	550
2	210	800
3	760	1150
4	1010	1600
5	1515	2200
6	2010	3200
7	3150	4500
8	4110	5500
9	5050	6500
10	6160	6600
11	6600	7500
12	7500	7300
13	7300	7600
14	7625	7900
15	7800	7980
16	8100	-

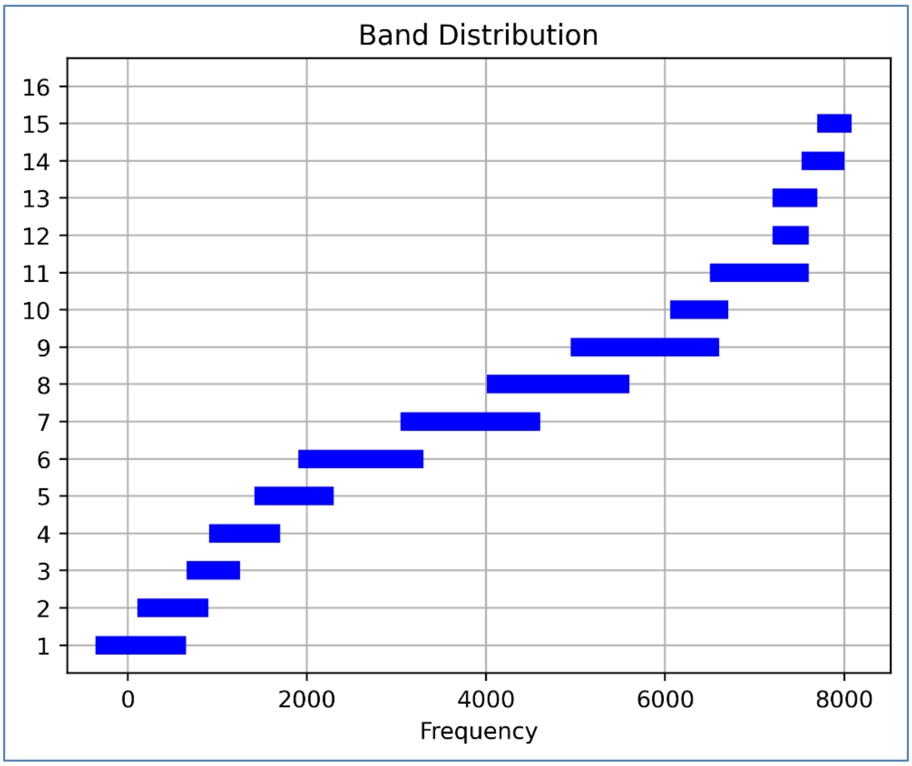

Fig 1: Representation of Frequency Band

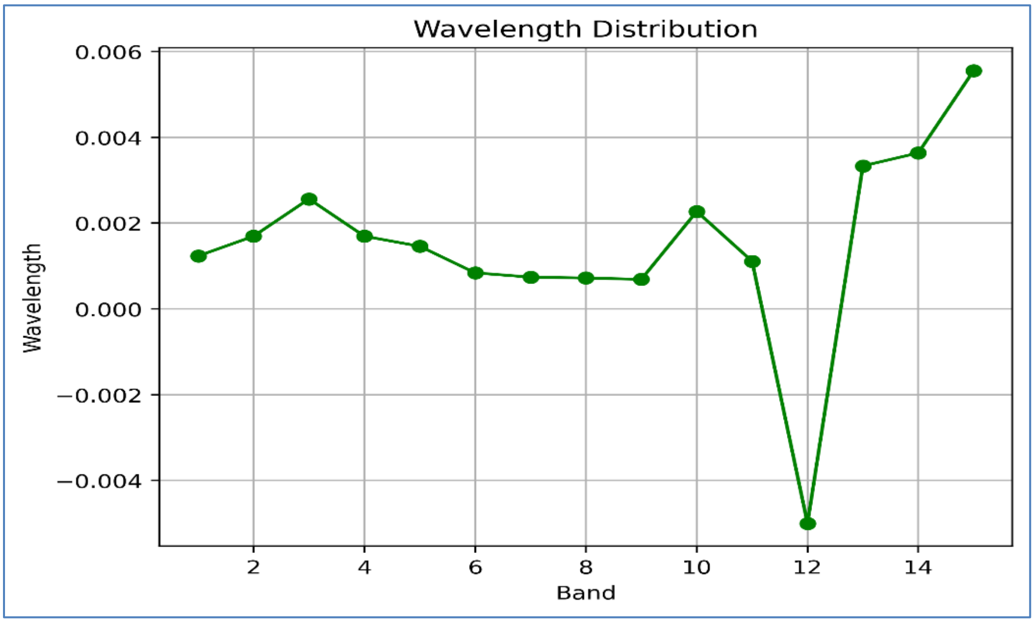

Fig 2: Distribution of Wavelength

B. Dynamic Range Multiband Compression

The diagram in figure 3 shows the proposed method. It visually outlines the step-by-step process of applying MDRC, starting with splitting the audio signal into frequency groups using digital signal processing [18]. It then independently adjusts the compression settings for each sub-group. The diagram serves as a thorough guide to grasping the details of the proposed method. Multiband Dynamic Range Compression is a customized and advanced approach for hearing aids. It addresses people's specific hearing needs by personalizing compre-ssion in different frequency bands. The addition of a MATLAB graphical user interface further improves how people interact. It makes changing the compression settings more accessible and intuitive. The outlined step-by-step proposed method offers a comprehensive grasp of the specifics required to apply MDRC for better hearing experiences.

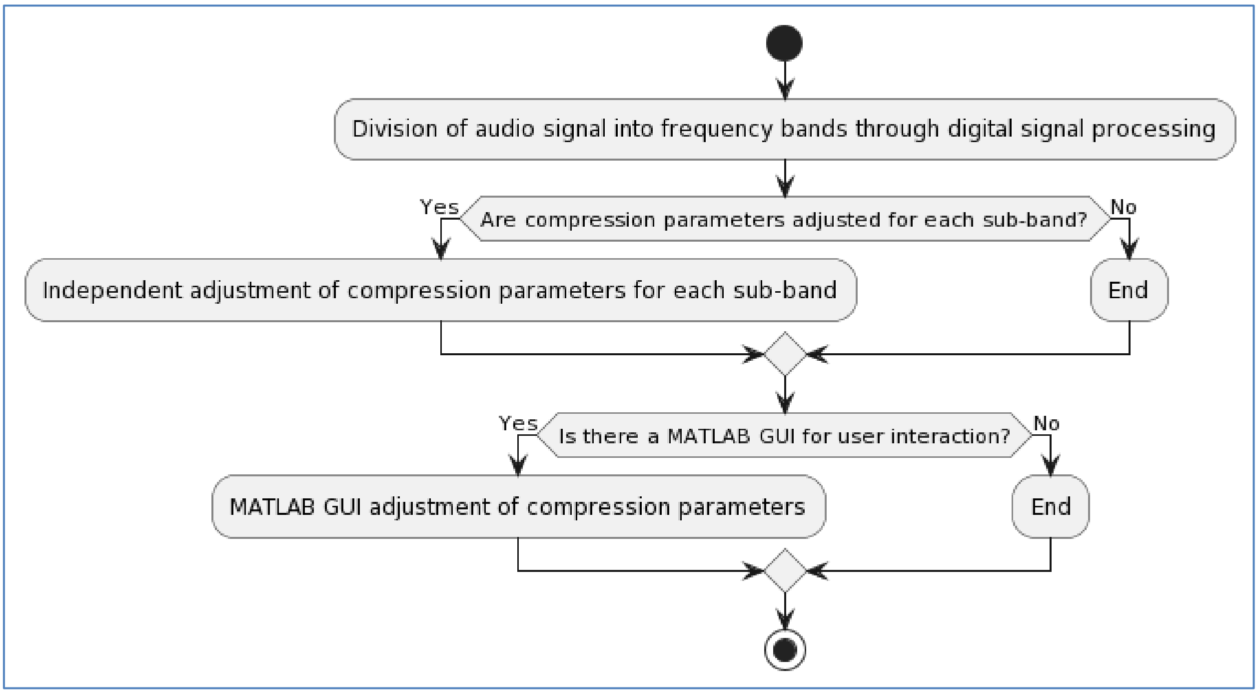

Fig 3: Representation of proposed method flowchart

Step 1: Multiband Dynamic Range Compression (MDRC):

Hearing aids use a technique called Multiband Dynamic Range Compression to help people with hearing loss better understand speech and other sounds. This method splits sound into different frequency ranges and compresses each one separately. It compresses loud sounds more than quiet ones so the full range of volumes stays usable. This aims to make hearing easier by processing different pitches independently. It provides a clearer listening experience than traditional hearing aids alone.

Step 2: Customization for Hearing Loss Patients:

MDRC helps individuals with hearing loss by adjusting how sounds are made louder in different pitches. Instead of making all sounds equally louder like regular hearing aids, MDRC can make higher or lower pitches louder than others. This customized adjustment helps make speech and other sounds clearer. It gives a balanced boost to sounds that may need more help for a person to hear better.

Step 3: Digital Signal Processing with Filter Bank:

MDRC breaks down sounds into different parts. It uses filters to separate the audio into high, medium, and low pitches. Then it can adjust how much each part is quieted specially. This gives a precise mix without losing important pieces.

Step 4: Independent Adjustment of Compression Parameters:

After separating the audio signal into parts, a special method compresses each section. A key part of this Multiple Band Loudness Adjustment is independently changing how much each part is compressed. The changes include the squash ratio, how fast it acts, how fast it lets go, and the volume threshold. Having separate control of each section lets compression traits be precisely set across different pitches. This detailed adjustment meets people's specific needs. Breaking the audio into parts means each piece gets its own compre-ssion method.

Step 5: MATLAB GUI for Parameter Adjustment:

The graphical user interface developed in Matlab lets people easily change how the sound is squeezed. Users can see and control things like how much the sound is squeezed, how fast it starts being squeezed, how fast it stops, and when squeezing begins. This visual tool improves how folks access and control the settings for squeezing the sound. It helps make the adjustments more focused on what people need.

4. Experimentation Results

A. Low-frequency Hearing Loss Result

Upon examination of Figure 5, the graph showing how loud different pitches sound allows us to see how the volume changes at different frequencies. Any weird spots, bumps, or lines in the low pitches can show how well the system helps with hearing low sounds. The picture makes it easy to see if the system can make soft low pitches louder like they should be to test how good it works. When testing for low pitch hearing loss, it's important to look closely at the frequencies below 1000 Hz. A well-made system would have a line showing the right amount of loudness in the low ranges so important noises like deep voices and bass notes can be heard better.

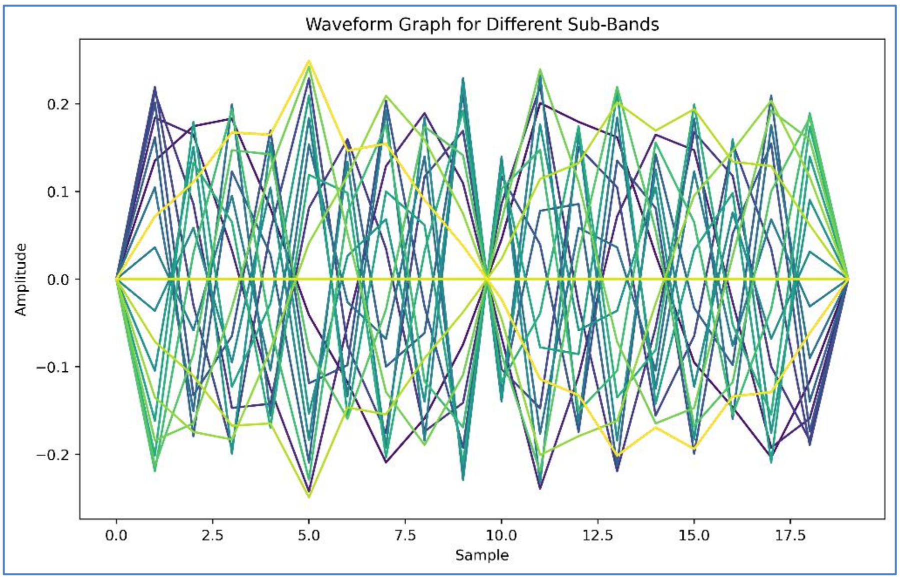

Fig 4: Sub Band Distribution for different waveform

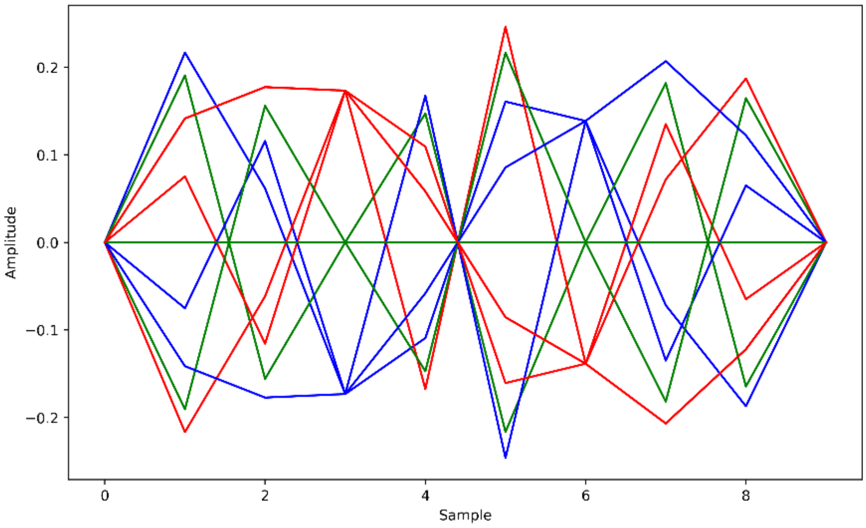

Fig 5: Waveform Graph for Different Sub-Bands

B. High-frequency Hearing Loss

It is crucial that we validate results for hearing loss affecting high pitches above 3000 Hz. This range includes distinct speech sounds like s's, high musical notes, and faint environmental noises like small birds. In our experiment, we focused on an audio sample with the word "please" spoken by women. Their voices were intentionally selected for containing many high tones, creating an ideal test for how well our system handles hearing loss problems in that range. The outcomes shown on the user interface (UI) in Figure [insert figure number] provide a clear picture of how our system performs when helping with high-frequency hearing loss. By reviewing the UI results, we can understand how effectively the system compensates for missing high tones in the chosen audio. This evaluation specifically matters for people having trouble hearing high pitches, and the UI serves as a visual guide to judge the system's success.

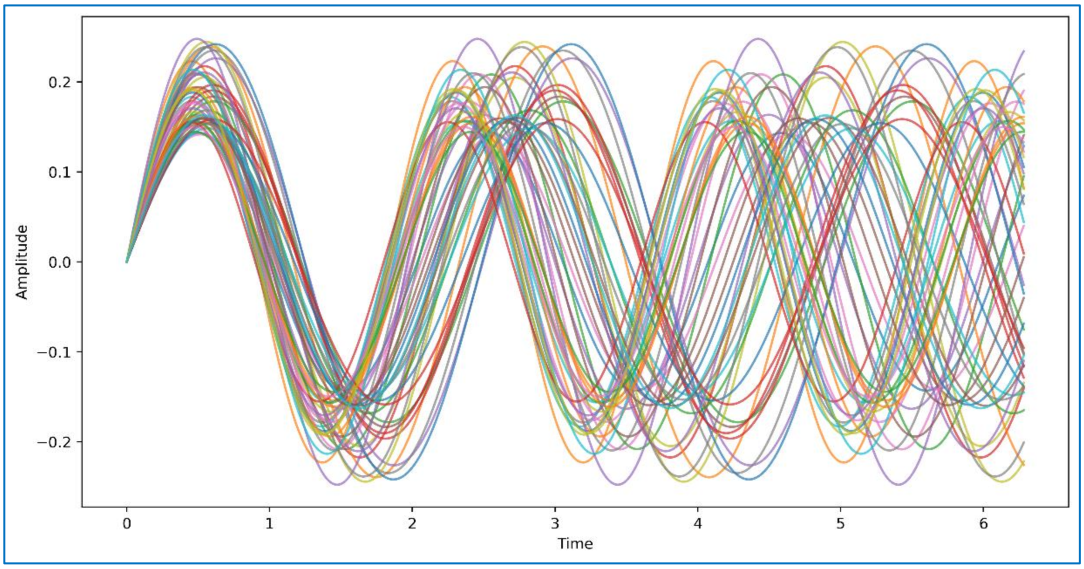

Fig 6: Representation of Waveform for High-Frequency Hearing Loss Results

In addition, using an audio sample with many high pitches makes it more important to test how well people hear, as it shows what real life is like when some have a hard time hearing certain small speech or environmental sounds. The pictures from the computer program in Figure [insert figure number] give a complete look at the results, letting us better grasp how well the system can boost what people can hear and understand in the higher sounds.

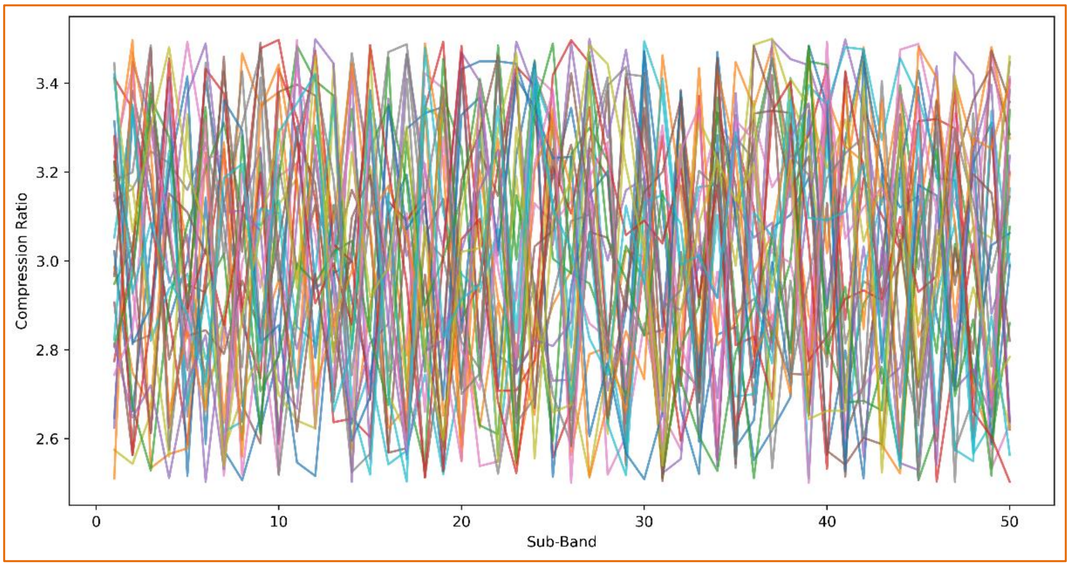

Fig 7: Subband Distribution of High-Frequency

The experiment helped test how well the system works for people with high frequency hearing loss. Using a sound clip that mattered and showing the results helped explain what the system can do for problems hearing high pitches. It was good to pick something important to listen to and use pictures to show what the machine le-arned.

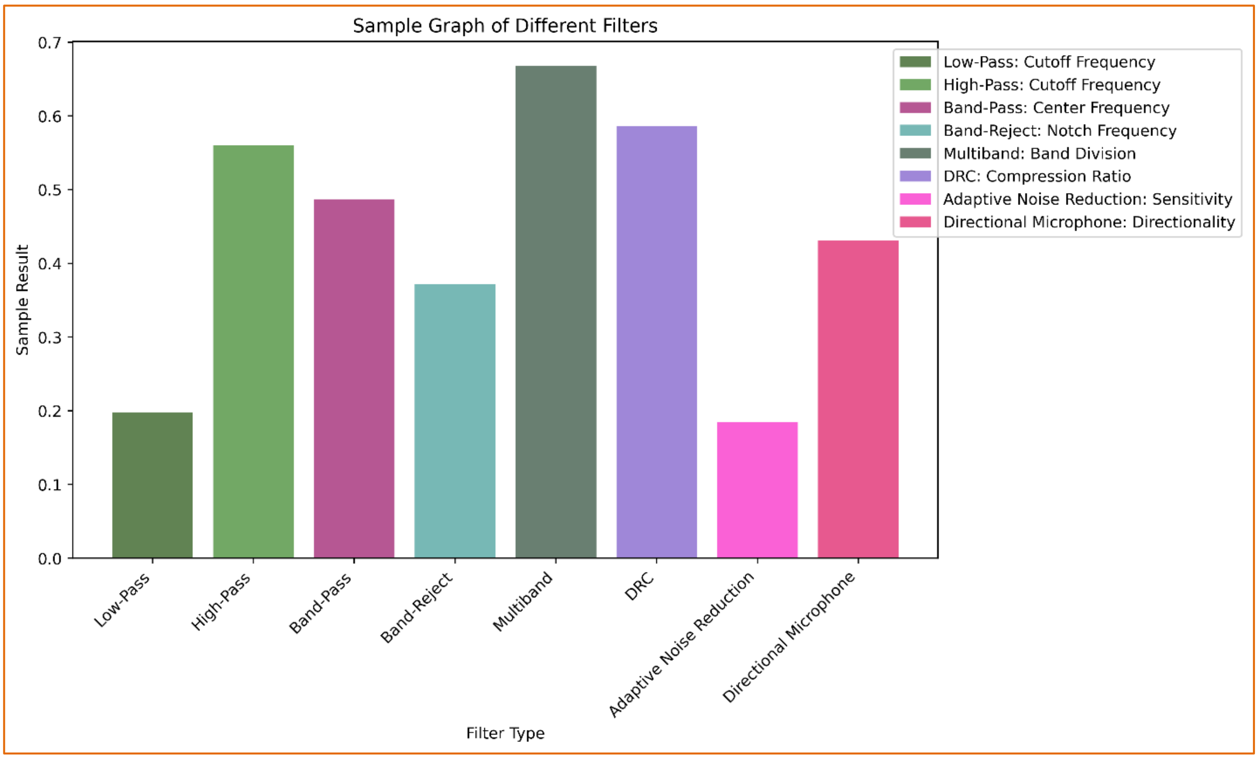

Fig 8: Representation of different filter in for Hearing Aid

5. Conclusion

The proposed Dynamic Range Compression (DRC) Filterbank with Multiband Adaptability for hearing aids presents a promising solution to address the different needs people have with hearing problems. It uses Freque-ncy Response Masking (FRM) in how the filterbank is made which allows the sounds to be shared out in a way that can be changed. This helps the filterbank work with each person's own way of hearing. The Multiband Dynamic Range Compression (MBDRC) technique makes it more able to change by adjusting how sounds are made louder or softer for different groups of pitches. This gives a plan for making sounds louder that is made for each person. Putting a Graphical User Interface (GUI) in the program MATLAB helps people use it more. It makes it easy to see and change how loud sounds are made and how the groups of pitches work. Looking at everything together, from how the filterbank is made to using multiband dynamic range compression, shows a full solution to help people with hearing loss hear better. The system works well because it can make sounds louder in certain pitches to help understand speech and make sounds better across different groups of pitches. This research is a big step forward in hearing aid technology. It focuses on making things able to change, made for each person, and easy to use to help people with different hearing problems hear better.

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