# Filter Bank Realization of Discrete Wavelet Transform based Multiband Dynamic Range Control for Audio

Lavanya C* and Dr. Arvind H.S**
* Research Student, M.Tech, B.E, E&C Dept, Bangalore, India
Email: lavanya.kites@gmail.com
** Professor&HOD, E&C Dept, JSS Academy of Technical Education, Bangalore, India
Email: aravindhs1@gmail.com

**Abstract:** This paper proposes a one-level one dimensional Discrete Wavelet Filter bank based Multiband Dynamic Range Control DRC for audio. Using different basis functions like daubechie, haar and dmeyer DWT decomposes input audio signals into low and high frequency bands with the help of analysis filters, and these sub-bands are decimated by a factor of 2 and inputted to DRC up-sampled by a factor of 2 after which need to be combined to reconstruct the original signal with the help of synthesis filters. The reconstructed signal is an exact replica of input signal called perfect reconstruction with better sound quality, very less computational time and best frequency response when compared to QMF filter bank. In DRC, multi-band compressor is used to apply compression differently to different frequency bands of the input signal, which uses minimum gain method. This allows the user to be selective about how compression is applied to a signal and only add power to certain parts of the frequency spectrum. Here limiter, compressor, expander and noise gate are used in the DRC, which protects the AD converter from overload.

**Keywords**: Dynamic Range Control (DRC), Discrete Wavelet Transform (DWT), Quadrature Mirror Filter (QMF), Analog to digital converter (A/D).

## Introduction

For fast and efficient implementation of DWT transform is used for filter implementation of dynamic range control. Wavelet computation is periodic with N samples, the maximum number of filters needed for computation in a one-dimensional multilevel forward wavelet transform is two. In other words, one low-pass and one high-pass filter will always be adequate for computation of 1-dimensional DWT. DWT can separate frequential components with higher frequency resolution co-efficients, as the frequency increases compared to Fourier transform. This means that at bigger frequencies, the number of components that can be distinguished is larger. It used in signal processing applications for separating signals into sub-bands and reconstructing them from individual sub-bands using down and up-samplers [1]. Advantage of this architecture is that the word length can be arbitrary and uses basis functions that are localized in both the real and Fourier space.

The dynamic range of a signal is defined as the logarithmic ratio of maximum to minimum signal amplitude and is given in decibels. The combination of level measurement and adaptive signal level adjustment is called dynamic range control. While recording, dynamic range control protects the A/D converter from overload or is employed in the signal path to optimally use the full amplitude range of a recording system. For suppressing low-level noise, so called noise gates are used so that the audio signal is passed through only from a certain level on wards [2]. Our result demonstrates the performance and benefits of using DWT filter bank along with DRC algorithm against using only QMF filter bank for DRC algorithm using MATLAB software.

## Dynamic Range Control

DRC provides compression and expansion capabilities that allow audio signals to sound softer or louder depending on the input signal protecting A/D converter from overload. It is necessary to modify broadcast signal because the channel has defined peak limit at which severe distortion and overload can occur and lower limit determined with noise.

DRC has the following components: limiters, compressors, expanders, noise gates, attack and release time calculation and smoothing. With the help of a limiter, the output level is limited when the input level exceeds the limiter threshold LT. The compressor maps a change of input level to a certain smaller change of output level. The expander increases changes in the input level to larger changes in the output level. The noise gate is used to suppress low-level signals, for noise reduction and also for sound effects like truncating the decay of room reverberation. This helps in controlling the transient attack of percussive instruments such as drums, raising the over- all loudness of a sound source by applying compression with make-up gain and providing a more consistent signal level, thus modifies dynamic range without introducing distortions [3]. These dynamic characteristics are shown in Fig:2.

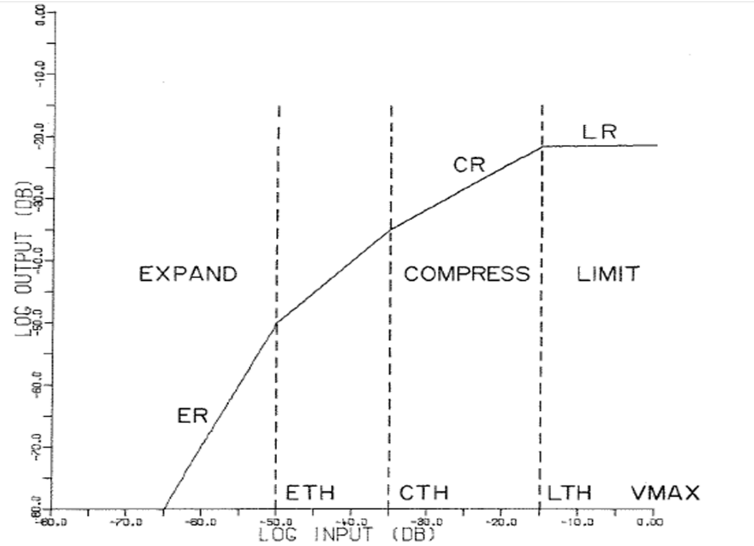


Fig.1 Input Characteristics of DRC with the parameters (LR = limiter Ratio, CR= compressor Ratio, ER = expander Ratio)

In the logarithmic representation of the static curve the compression factor R ratio is defined as the ratio of the input level change $\Delta P_I$ to the output level change $\Delta P_O$:

$$R = \frac{\Delta P_1}{\Delta P_0} \qquad (1)$$

With the help of Fig.1 straight line equation and the compression factor as:

$$Y_{dB}(n) = CR + R{-}1(X_{dB}(n) - CR) \qquad (2)$$

$$R = \frac{X_{dB}(n)\text{-CR}}{Y_{dB}(n)\text{-CR}} = \tan \beta_C \qquad (3)$$

are obtained, where the angle $\beta$ is defined as shown in Fig.1.

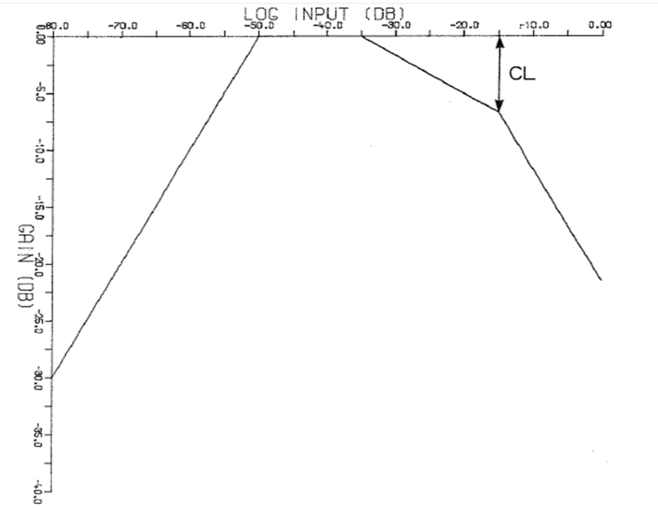


Fig.2: DRC Gain Characteristics with the parameters (LT = limiter threshold, CT= compressor threshold, ET = expander threshold and NT = noise gate threshold)

limiter, $R > 1$ for compressor (CR: compressor ratio), $0<R < 1$ for expander, expander ratio $ER$, $R = 0$ for noise gate. Using Fig.1, the formulas for slopes and limits, which are used for calculation purpose, are obtained as [2],[3]:

*Compressor Slope R = (CR – PP) / CR.* (6)

*Expander Slope S = (ER– PM) / (ER – NR).* (7)

*Compressor Limit CL = (PP – CT –M) / R) + CR.* (8)

*Expander Limit EL = ER – ((ER – M - NT) / S).* *(9)*

**Peak and RMS measurement**

For PEAK measurement, the absolute value of the input is compared with the peak value. If the absolute value is greater than the peak value, the difference is weighted with the co-efficient AT attack time and added to (1-AT). $x_{PEAK}(n-1)$. For this attack case: $|x(n)| > x_{PEAK}(n-1)$ we get the difference equation and transfer function as[2],[3]:

$$x_{PEAK}(n) = (1 - AT).x_{PEAK}(n-1) + AT.|x(n)|\ . \quad (10)$$

$$H(z) = \frac{AT}{1-(1-AT)z^{-1}} \quad (11)$$

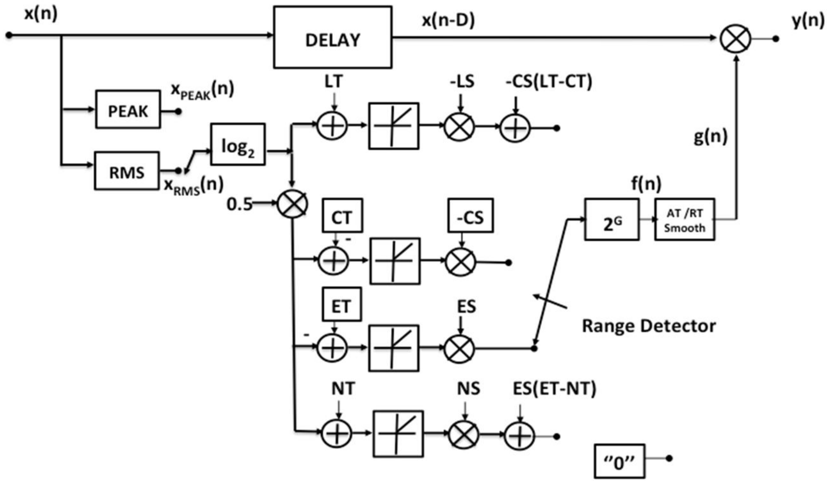


Fig.3: Block diagram for Dynamic Range Controller

If the absolute value of the input is smaller than the peak value for release case : $|x(n)| \le x_{PEAK}(n-1)$, the new peak value is given by :

$$x_{PEAK}(n) = (1 - RT).x_{PEAK}(n-1) \quad (12)$$

with the release time coefficient RT. The release case the transfer function is:

$$H(z) = \frac{1}{1-(1-RT)z^{-1}} \quad (13)$$

For the attack case the transfer function *H(z)* with coefficient *AT* and for the release case the transfer function *H(z)* with the coefficient *RT* is used. The coefficients are given by:

$$AT = 1 - exp\left(\frac{-2.2T_S}{t_a/1000}\right) \quad (14)$$

$$RT = 1 - exp\left(\frac{-2.2T_S}{t_r/1000}\right) \quad (15)$$

where, attack time $t_a$ and release time $t_r$ are given in milliseconds, sampling interval $T_S$ to achieve fast attack time response. The computation of the RMS value is done using:

$$x_{RMS}(n) = \sqrt{\frac{1}{N}\sum_{i=0}^{N-1} x^2(n-i)} \quad (16)$$

over N input samples can be achieved by a recursive formulation. The RMS measurement uses square of the input and performs averaging with a first-order low-pass filter. The averaging co-efficient is [4]:

$$TAV = 1 - exp\left(\frac{-2.2T_A}{t_M/1000}\right). \quad (17)$$

where $t_M$ *is* averaging time in milliseconds. The transfer function:

$$H(z) = \frac{TAV}{1-(1-TAV)z^{-1}}\ . \quad (18)$$

**Gain Factor Smoothing filter and Attack Time Measurement**

The difference function of gain factor smoothing filter is:

$$g(n) = (1 - k).g(n-1) + k.f(n). \quad (19)$$

and the corresponding transfer function leads to :

$$H(z) = \frac{k}{1-(1-k)z^{-1}} \ . \qquad (20)$$

The relationship between attack time ta and the time constant $\tau$ of the step response is obtained as follows [4]:

$$t_a = t_{90} - t_{10} = 2.2\tau \ . \qquad (21)$$

## DWT FILTER Bank Based Dynamic Range Control

In an L-level wavelet, the total number of samples computed is: 2(N-1). Here we are implementing 1-Level DWT filter bank. Since the wavelet computation is periodic with N samples, the number of samples computed every sample period is: [2(N-1)]/N, requires one low-pass and one high-pass filter as analysis and synthesis filters respectively gives approximation coefficients and detailed coefficients as their outputs , will always be adequate for computation of 1-dimensional DWT. For the DWT implementation here, the basis functions used are Daubechie, Haar and dmeyer. The generated DWT filter co-efficients $H_0$, $H_1$ are passed through analysis filters respectively and then are down-sampled and passed dynamic range control system and then its output is up sampled and passed through synthesis filters. Furthermore decomposing the low pass filter components even more to get higher resolution coefficients, i.e, The asymmetric structure of cascaded L-level DWT filter bank, the limitation here is that if the DWT coefficients of level L are of use, one has to first obtain the DWT coefficients at level L-1, thus increasing computational burden [5].

Here, presented an efficient one-dimensional direct DWT computation algorithm as shown in Fig.4. The algorithm enables computation of 1st-level DWT coefficients.

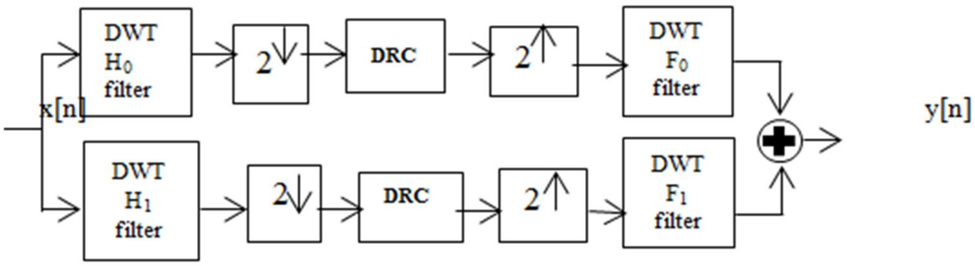


Fig.4: DWT Filter bank based Dynamic Range Control

The algorithm is simple and uses modified DWT filter structure generated out of basic analysis and synthesis filter bank. Here, low-frequency content of the signal is an important part compared to high-frequency content. In general, we can represent $G_J(z)$ the equivalent filter to the Jth stages of lowpass filtering and subsampling by $2^J$ [3]:

$$G^J(z) = \prod_{l=0}^{J-1} G(z^{2^l}) \qquad (22)$$

A necessary condition for the iterated functions to converge to a continuous limit is that the filter G(z) should have sufficient number of zeros at z = -1, or half sampling frequency, so as to attenuate repeat spectra [6]. Using this condition, the regular filters, which are both orthogonal and converge to continuous functions with compact support, may be generated. The well-known Daubechies orthonormal filters are deduced from maximally flat low-pass filters is mainly used.

The two fundamental conditions to give filters and wavelets with good properties, stated here for a two-channel filter bank, we use the polynomials $H_0(z)$, $H_1(z)$, $F_0(z)$, $F_1(z)$ whose coefficients come directly from DWT.

### Perfect Reconstruction Condition

The synthesis bank inverts the analysis bank, with L delays.

$$F_0(z)H_0(z) + F_1(z)H_1(z) = 2z^{-1} \qquad (23) \qquad \text{and } \textit{for anti} - \textit{aliasing choices},$$

$$F_0(z)H_0(-z) + F_1(z)H_1(-z) = 0 \qquad (24)$$

### Orthogonality Condition

The analysis bank is inverted by its transpose. The wavelets are orthogonal to all its dilates and translates. The filter coefficients are reversed by:

$$F_0(z) = z^{-N}H_0(z^{-1}) \ and \ F_1(z) = z^{-N}H_1(z^{-1}) \quad (25)$$

$$\sum_{n=0}^{N} h[k]\, h[k+2n] = \delta(n) \qquad (26)$$

## Simulation Results for Dynamic Range Control Using different DWT filter banks

***Design Example 1:*** Assumed parameters for DRC are: ATime = 2000e$^{-6}$ is Attacking time, RTime = 4000e$^{-6}$ is Release time, TAVime =6000e$^{-6}$ is Averaging time [8].
The coefficients attack time AT and release time RT are given by are:
AT=1 - exp(- 2.2*sampling time / ATime),
RT=1- exp(- 2.2*sampling time / RTime),
TAV = 1 - exp(- 2.2*sampling time / TAVime).
Assumed values during simulation are: Compressor Threshold CT= -40, Expander Threshold ET= -50, Constant Gain M=12, Noise Threshold NT= -80, Peak Power PP= -5, Minimum Power PM= -100.
The compressor/expander operates in the linear region of the static curve if the control factor is equal to 4. If the control factor is between 1 and 4. The system operates as a compressor. For control factors lower than 1, the system works as an expander ($3500<n<4500$ and $6800<n<7900$). The compressor is responsible for increasing the loudness of the signal [7], whereas the expander increases the dynamic range for signals of small amplitude. Fig4: shows the DRC gain ouptut for input audio signal which varies between 2 to 4.

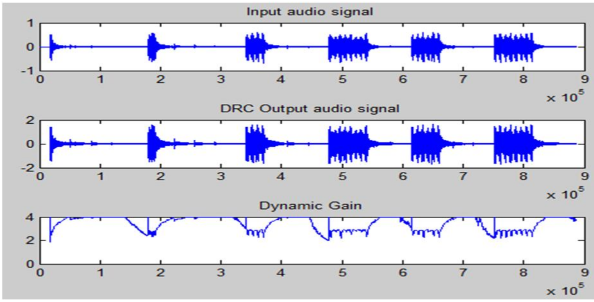


Fig.5: DRC and dynamic gain are shown for input audio signal without using any filters

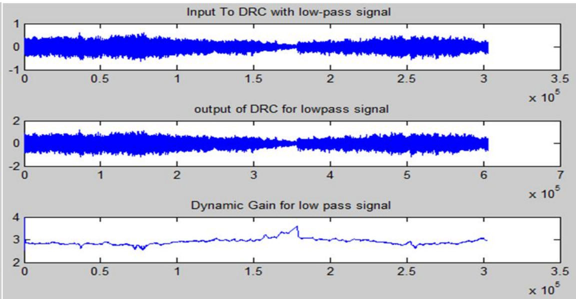


Fig.6 Input to DRC and output from DRC for lowpass signal and Dynamic gain using Daubechies wavelet filter bank

The selected 1-Level DWT filter bank is a Daubechies filter with wavelet order 10. The output from DRC is sent to synthesis filters (high-pass synthesis FIR filters) with IDWT and output is shown in Fig.6. The DRC gain ouptut for input audio signal which varies between 0 to 4,hence the range is controlled with better sound quality when compared with QMF filter bank and takes less computational time. In Fig.7, for Haar wavelet filter, the DRC gain ouptut for input audio signal which varies between 2 to 4 for highpass signal, rest for lowpass signal dynamic gain remains same as daubechie filter bank. Hence, daybechie filter bank is the best selection.
Fig.9, 10&11: is an frequency response plot of generated DWT based Daubechies, Haar and dmeyer wavelet filter banks. It is evident from the plots that, the output signal is completed reconstructed only in daubechies filter whose frequency responses exactly matches ideal filter response. This again confirms suitability of daubechie filter structure for DWT decomposition. The output audio is compared with input audio and there is an increase in its amplitude with improved quality by using DWT filter bank and mainly the processing speed is increased and out of all wavelet filter banks, output signals are present in passband of Daubechies filter which is the best suitable wavelet filter bank as shown in Fig.9 when compared to QMF filter response.

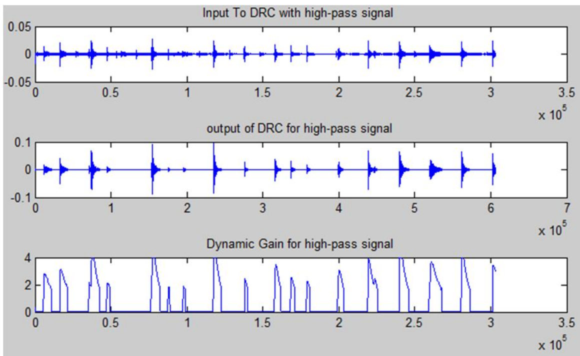


Fig.7 Input to DRC and output from DRC for high-pass signal and Dynamic gain gain using Daubechies wavelet filter bank

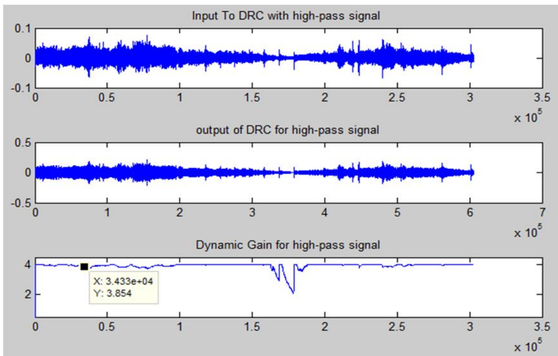


Fig.8 Input to DRC and output from DRC for highpass signal and Dynamic using Haar wavelet filter bank

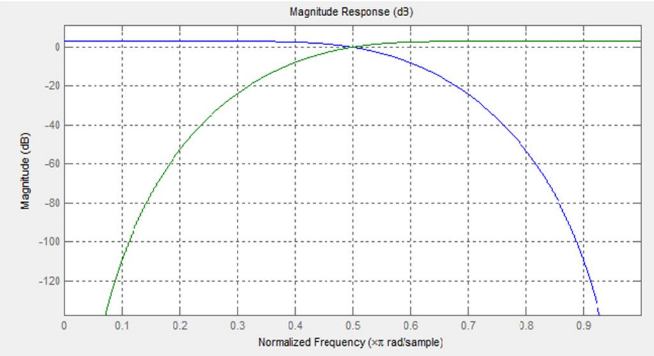


Fig.9: Magnitude response for DWT based Daubechies wavelet filter bank

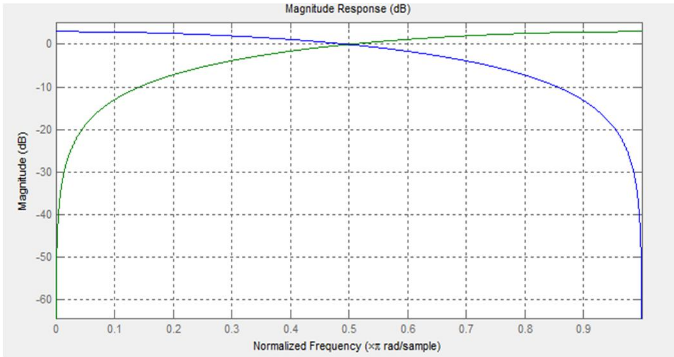


Fig.10: Magnitude response for 2-channel DWT based Haar wavelet Haar wavelet filter bank with wavelet order 10

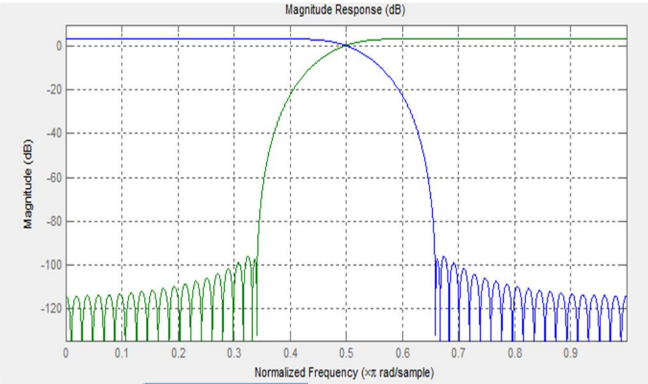


Fig.11: Magnitude response for 2-channel DWT based dmeyer wavelet filter bank

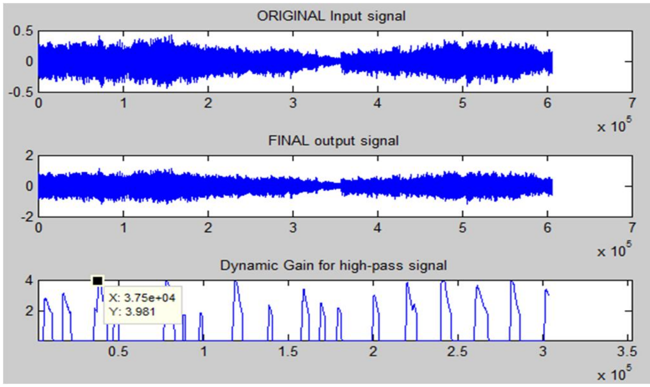


Fig.12 Input audio and final output audio signals with dynamic gain

Derived values are as follows: CT = -7, ET = -50 and NT = -80. We observed that for constant gain M equal to 12dB, there is a shift for dynamic gain equal to G equals 3.981 from Fig:12, practically which matches the theoretical value 4 approximately and hence dynamic range of audio is controlled by using one-level DWT filter bank with better sound quality and less computational time when compared to QMF filter bank.

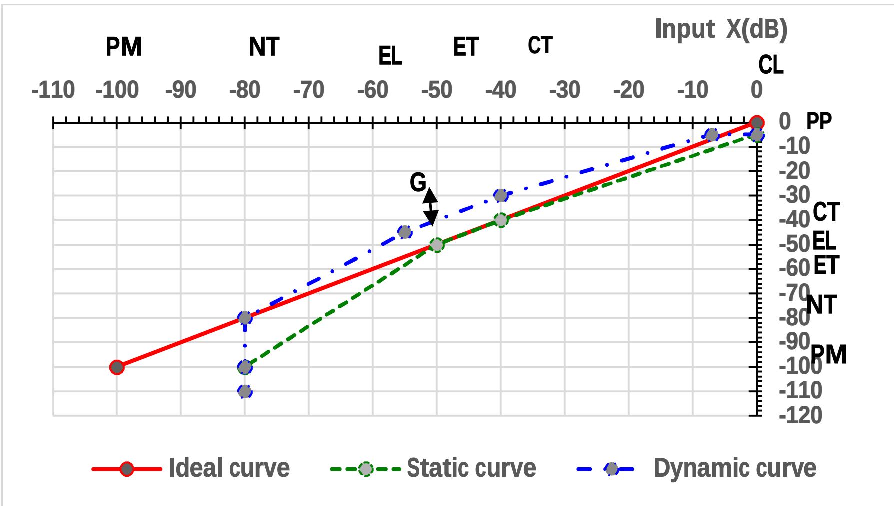


Fig.12: Ideal, Static and Dynamic Gain curve for output audio signal

## Conclusions

In this paper, we observed that, the comparison of QMF filter bank and other wavelet filter banks with the daubechie filter bank structure is analysed. The computational time for DWT required is very less with reduced filter length. The quality of output audio is very clear considering all most frequency components with efficient filter coefficients. Hence, Daubechie filter bank structure is well suited for Dynamic range control with improves the audio quality and hence protects the analog to digital converter by controlling the dynamic range. An alternative structure of DWT in terms of parallel filters is also derived. Impulse response and frequency response plots of generated parallel filter structure validate its suitability in terms of dyadic frequency selectivity.

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