IOSR Journal of Electronics and Communication Engineering (IOSR-JECE)
e-ISSN: 2278-2834,p- ISSN: 2278-8735.Volume 9, Issue 3, Ver. IV (May - Jun. 2014), PP 19-24
www.iosrjournals.org

Comparison of Multirate two-channel Quadrature Mirror Filter Bank with FIR Filters Based Multiband Dynamic Range Control for audio

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Abstract: *This paper proposes a Comparison of Multirate two-channel Quadrature Mirror Filter QMF bank with FIR filters based Multiband Dynamic Range Control DRC for audio. Two-channel QMF 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 with increased amplitude in the audio compared to when only FIR filters are used. The filters used here are Finite Impulse Response FIR using Kaiser Window. The reconstructed signal is an exact replica of input signal with some delay called perfect reconstruction. 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.*

Index Terms: *Analog to digital converter (A/D), Dynamic Range Control (DRC), Finite Impulse Response (FIR), Quadrature Mirror Filter (QMF).*

I. Introduction

Among the various filter banks, two-channel QMF bank was the first type of filter bank used in signal processing applications for separating signals into sub-bands and reconstructing them from individual sub-bands using down and up-samplers. It uses the full amplitude range of a recording system for multiband audio signals.

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 [1]. Our result demonstrates the performance and benefits of using QMF filter bank along with DRC algorithm against using only DRC algorithm using MATLAB software.

II. Dynamic Range Control

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 [2].

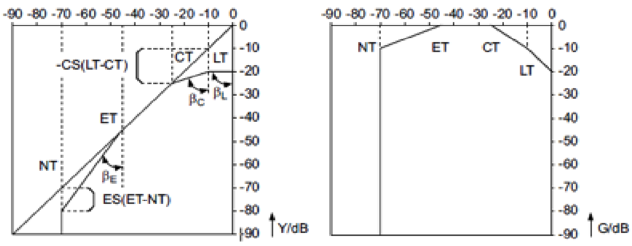

Fig.1 Static Characteristics of DRC with the parameters.

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 ΔP_O:

$$R = \frac{\Delta P_1}{\Delta P_0} \tag{1}$$

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

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

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

are obtained, where the angle β is defined as shown in Fig.1.
The relationship between ratio R and the slope S can also be derived from Fig.1 and is expressed as:

$$S = 1 - \frac{1}{R} \tag{4}$$

$$R = \frac{1}{1-S} \tag{5}$$

The block diagram for Dynamic Range Control is shown in Fig.2. Typical compression factors are: Slope $R=\infty$ for 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:

Compressor Slope $R = (CT – PP) / CT$ (6)

Expander Slope $S = (ET – PM) / (ET – NT)$ (7)

Compressor Limit $CL = (PP – CT –M) / R) + CT$ (8)

Expander Limit $EL = ET – ((ET – M - NT) / S)$ (9)

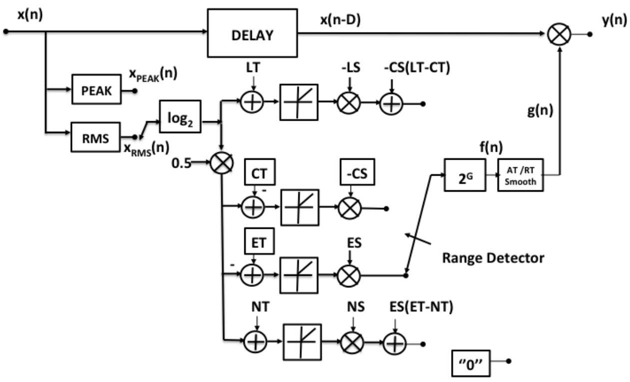

Fig.2 Block diagram for Dynamic Range Controller

A. Peak and RMS measurement

For PEAK measurement, the absolute value of the input is compared with the peak value. If the absolute value is greate rthan the peak value, the difference is weighted with the co-efficient *AT* attack time and added to *(1-AT). xPEAK(n-1).*

For this attack case: $|x(n)| > xPEAK(n-1)$ we get the difference equation and transfer function as[2]-[3]:

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

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

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

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

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

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

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 difference equation and the transfer function is given by:

$$x_{RMS}^2(n) = (1 - TAV).\, x_{RMS}^2(n-1) + TAV.\, x^2(n) \quad (18)$$

$$H(z) = \frac{TAV}{1-(1-TAV)z^{-1}} \quad (19)$$

B. Gain Factor Smoothing filter and Attack Time Measurement

The difference function of gain factor smoothing filter and the corresponding transfer function leads to,

$$g(n) = (1 - k).\, g(n-1) + k.\, f(n) \quad (20)$$

$$H(z) = \frac{k}{1-(1-k)z^{-1}} \quad (21)$$

With the definition of attack time $ta = t_{90} - t_{10}$, it follows as:

$$0.1 = 1 - e^{-\frac{t_{10}}{\tau}}. \quad \leftarrow t_{10} = 0.1\tau \quad (22)$$

$$0.9 = 1 - e^{-\frac{t_{90}}{\tau}}. \quad \leftarrow t_{90} = 0.9\tau \quad (23)$$

The relationship between attack time ta and the time constant τ of the step response is obtained as follows:

$$0.9/0.1 = e^{(t_{90}-t_{10})/\tau} \quad (24)$$

$$l\,n(0.9/0.1) = t_{90} - t_{10})/\tau \quad (25)$$

$$t_a = t_{90} - t_{10} = 2.2\tau \quad (26)$$

III. Two-Channel Quadrature Mirror Filter Bank Based Dynamic Range Control

Quadrature Mirror Filter QMF are called so since input signal x[n] is first passed through two-band i.e., multiband analysis FIR filters typically low and high-pass filters with cut-off frequency at $\pi/2$, corresponds to one fourth the sampling frequency. It is two-channel filter bank. The relation between the output and input of this system is expressible as [5]:

$$Y(z) = T(z)X(z) + A(z)X(-z) \quad (27)$$

$$T(z) = 1/2[H_0(z)F_0(z) + H_1(z)F_1(z)] \quad (28)$$

$$A(z) = 1/2[H_0(-z)F_0(z) + H_1(-z)F_1(z)] \quad (29)$$

are the distortion transfer function and the aliasing transfer function, respectively. The second term can be made zero by selecting the synthesis filters as $G_0\,(z) = 2H_1\,(-z)$ and $F_1(z) = -2H_0(z)$. In this case, the residual filter bank distortion becomes:

$$T(z) = H_0(z)H_1(-z) - H_1(z)H_0(-z) \quad (30)$$

We use the following transfer functions, instead of $H_0(z)$ and $H_1(z)$as follows,

$$G_1(z) = \sum_{n=0}^{N_1} g_1[n]\, z^{-n} \equiv H_1(-z)\sum_{n=0}^{N_1}(-1)^n h_1[n] \quad (31)$$

$$G_0(z) = \sum_{n=0}^{N_0} g_0[n]\, z^{-n} \equiv H_0(z) = \sum_{n=0}^{N_0} h_0[n]\, z^{-n} \quad (32)$$

In the following transfer functions $G_0(z)$ and $H_0(z)$ low-pass filters are identical, where as $G_1(z) = H_1(z)$. Therefore, $|G_1(e^{j\omega})| = |H_1(e^{j(\pi-\omega)})|$ so that the amplitude response of $G_1(z)$ is obtained from that of $H_1(z)$ by means of the substitution $(\pi - \omega)$ for ω and vice versa [6].

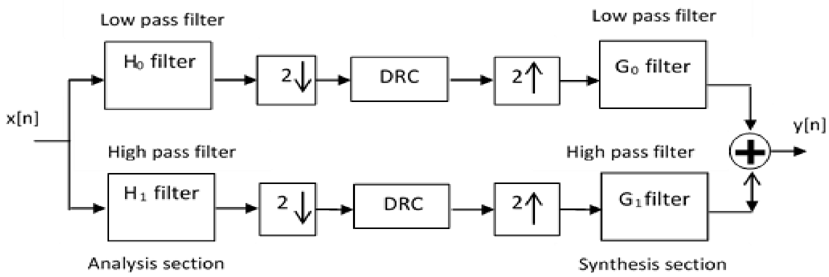

Fig.3 Two-channel Quadrature Mirror Filter based Dynamic Range Control

IV. Simulation Results for Dynamic Range Control Without And With Using QMF Filter Bank

Example 1: Figure 3 shows the DRC for input audio signal. Assumed parameters for DRC are: ATime = 2000e^{-6} is Attacking time, RTime = 4000e^{-6} is Release time, TAVime =6000e^{-6} is Averaging time.

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.

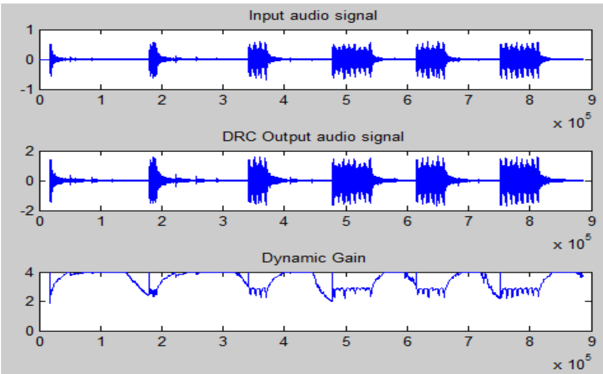

Fig.4 DRC and dynamic gain are shown for input audio signal without using any filters.

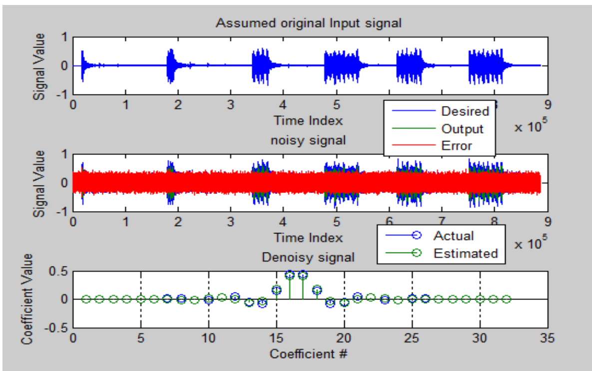

Fig.5 Input audio, noisy audio and de-noisy audio signals using filters.

Fig 4. Shows input, output signal and its dynamic gain for audio without using filters with dynamic gain equal to 4. Fig.5 shows the input audio signal and the noise that is present in the input is random noise with the parameters order of the FIR filter n is 31, normalized cut-off frequency w_n is 0.5, order of adaptive filter used to filter out the noise from input audio n is 32.The input from analysis filters (high-pass and low-pass FIR) with Kaiser windowing technique and is sent to DRC and output is shown in Fig.6 and Fig.7. The order of the filter taken is 99 and number of samples taken at a time during interpolation is 2.The order of the filter n taken is 99 and number of samples taken at a time for interpolators is 2. Two channel Quadrature Mirror effect i.e., $\pi/2$ corresponds to one fourth the sampling frequency is shown for high and low-pass FIR filters decimators and interpolators are shown in Fig.8. The order of the filter taken is 99 and number of samples added at a time for interpolators is 2.The output audio is compared with input audio and there is an increase in its amplitude with improved quality by using QMF filter bank and mainly the processing speed is increased as shown in Fig.9.

The output response from low and high-pass FIR filters is shown in Fig.11 and Fig.12, using Hanning window technique returns coefficients b with length $n+1$ with pass-band frequency fp = 3400 and 8000, f = 8000 and 44100, stop-band frequency fs = 3800 and 7700 respectively, order of multiband filter n = 111, cut-off frequency w_n ranges from w_1 = 500, w_2 = 8000 and is sent to DRC and output is shown in Figure 6.3. The order of the filter taken is 99 and number of samples taken at a time during interpolation = 2. Fig.13 shows the input and output audio for DRC using FIR filters only where its observed that the amplitude of the signal is less compared to input. Fig.15 is plotted using the assumed and derived values.

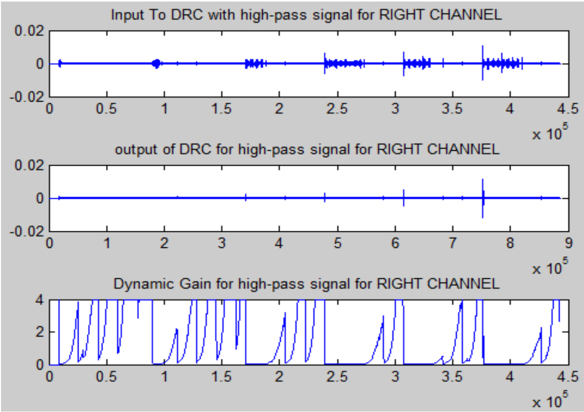

Fig.6 Input to DRC and output from DRC and Dynamic gain for Right channel using high-pass FIR filter (Decimators).

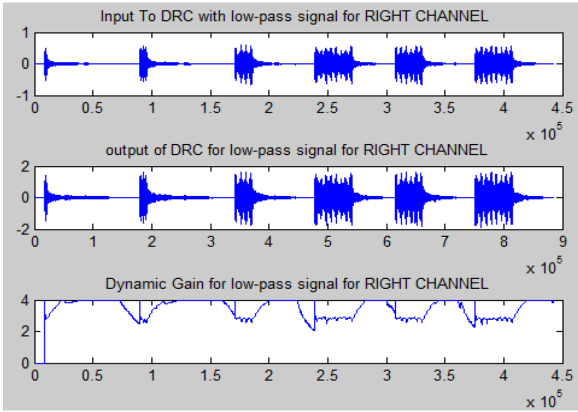

Fig.7 Input to DRC and output from DRC and Dynamic gain for Right channel using low-pass FIR filter (Decimators).

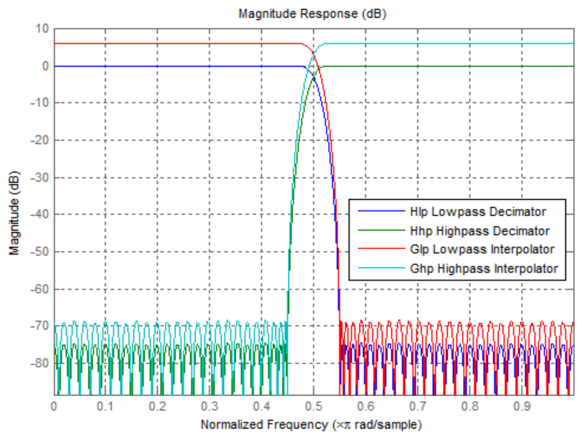

Fig.8 Magnitude response for 2-channel QMF Filter Bank.

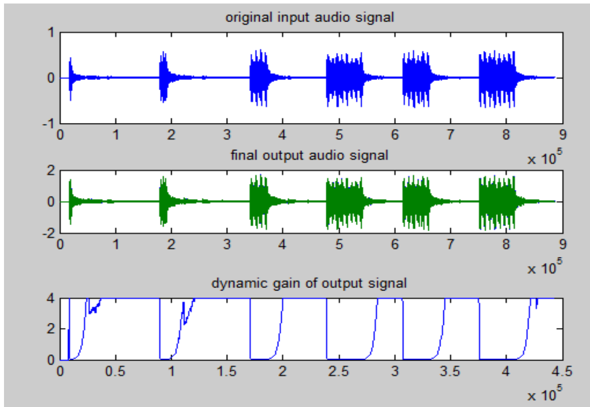

Fig.9 Input audio and final output audio signals.

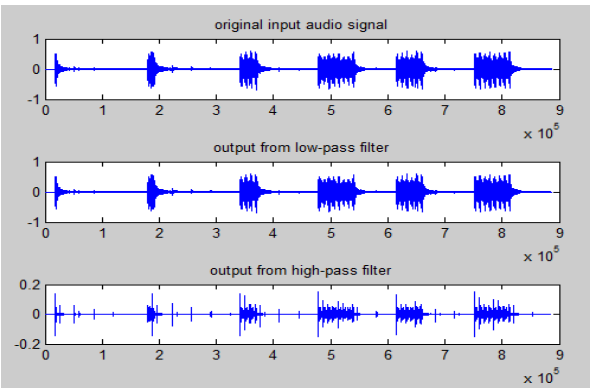

Fig.10 Input audio, output signal from low-pass and high -pass FIR filters.

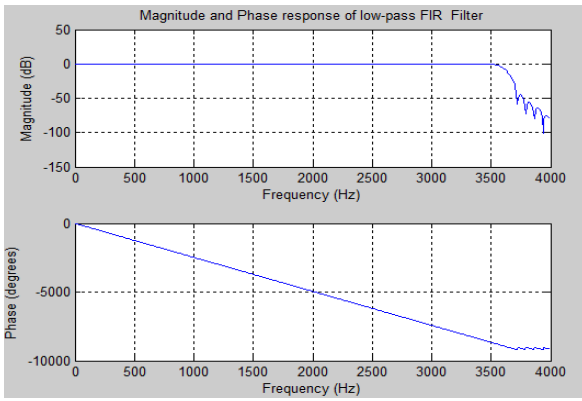

Fig. 11 Magnitude and Phase response for low-pass FIR filter after input audio is passed through it.

Hence, Multiband DRC using high and low-pass FIR filters protects the analog to digital converter by controlling the dynamic range. The use of multiband FIR filters increase the processing speed as takes only few samples as they are divided into low and high-pass filters but dynamically controlled output signal amplitude is reduced though dynamic gain G = 4 same as if QMF is used. Hence, this project proposes two-channel Quadrature Mirror filter bank for multiband dynamic range control instead of only using FIR filters.

Assumed values are as follows: Compressor Threshold *CT*= -40, Expander Threshold *ET*= -50, Constant Gain *M*=12, Noise Threshold *NT*= -80, Peak Power *PP*= -5, Minimum Power *PM*= -100. 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 3.98 practically which matches the theoretical value 4 approximately and hence dynamic range of audio is controlled by using two-channel QMF with DRC.

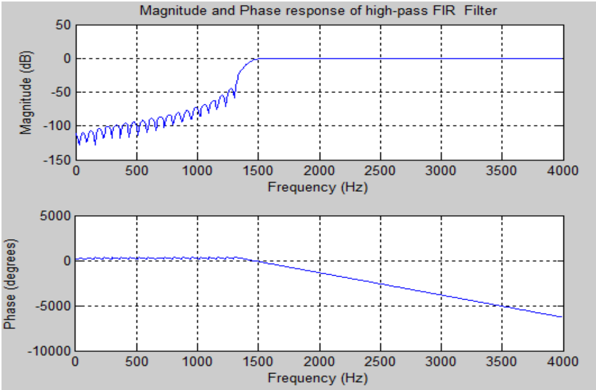

Fig. 12 Magnitude and Phase response for low-pass FIR filter after input audio is passed through it.

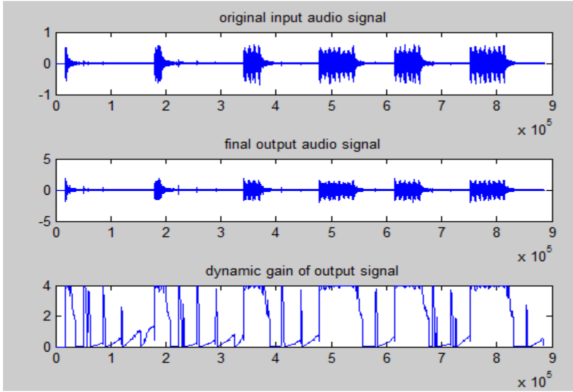

Fig. 13 Input audio, final output audio signals dynamic gain of final output signal

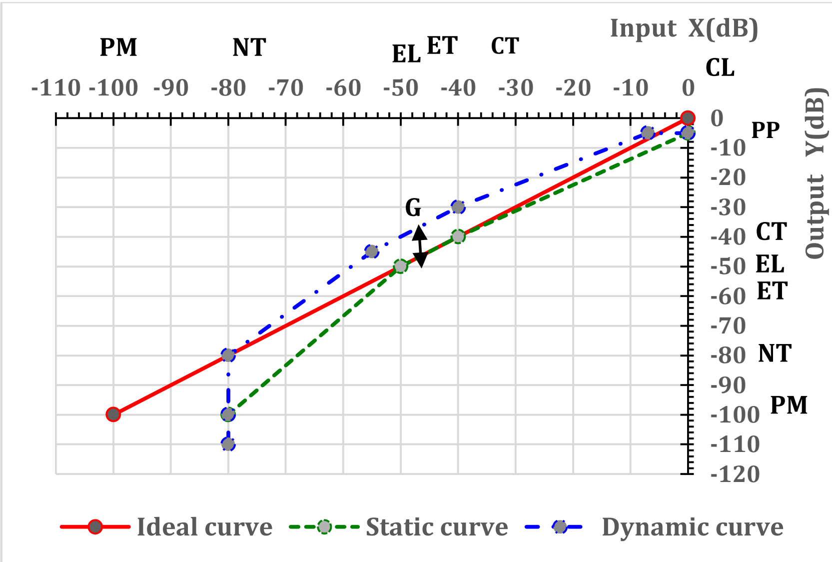

Fig.14 Ideal, Static and Dynamic Gain curve for output audio signal.

V. Conclusions

In this paper, it's observed that DRC using two-channel QMF filter bank, improves the audio quality with increase in amplitude and hence protects the analog to digital converter by controlling the dynamic range. The use of two-channel QMF filter bank up to sampling rate 4 increases the processing speed as interpolators takes only few samples and maintains the perfect reconstruction after decimation with small delay and matches the theoretical value for dynamic range control than FIR only. Kaiser window technique used in FIR filter design as it allows adjustment of the compromise between the overshoot reduction and transition region width spreading.

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