Estimation of Attack Time Constant for Dynamic Range Compressors in Hearing Aids

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***Abstract*—Dynamic Range Compression (DRC) is a key component in all modern Hearing Aids. Attack and Release time constants decide the speed with which the DRC should act to the incoming signal amplitude variation. So an accurate estimation of time constants gives a precise control over the DRC behavior. In this paper we examined various errors that occur in the output of the DRC while using conventional methods which affect attack and release time constants adversely. New methods are proposed for a better estimation of time constants in DRC. Since all the modifications are made in the estimation of attack time, there is no need to change the existing hardware for DRC. The proposed algorithm gives perfect output characteristics with zero error for test signals defined in ANSI S3.22 standards for hearing aid specifications.**

***Index Terms*—Hearing Aids; Dynamic Range Compression; Attack Time; Release Time; Non linear filtering**

I. Introduction

The dynamic range of audio signals that can be perceived by hearing impaired persons is lesser compared to persons with normal hearing. Dynamic range compression algorithm helps in reducing the high input level above a particular threshold value while keeping the softer sounds as it is. DRC algorithms are originally derived from general audio processing applications where slight variations in dynamic parameters may not make much difference in the final output since the people listening to those applications would be having normal hearing. But in the case of hearing impaired people a precise control over time constants is preferred, because even small variations in time constants may affect the speech quality and intelligibility [1]. Even 2 to 3 ms difference in time constants may degrade the spectral contrast especially in fast acting multiband dynamic range compressors [2].

Even though DRC is one of the widely used techniques in audio transmission and audio processing for reduction of channel bandwidth in communication and for artistical enhancement of music, there is not much information available about DRC theory. Blesser [3] discussed the importance of attack and release times in speech intelligibility. The fundamental theory behind digital DRC is explained in [4]. A smoothing stage was proposed to reduce the quantization errors that may occur during the hardware implementation of the algorithm. The author ignored the effect of gain calculation stage on time constant and followed input Sound Pressure Level (SPL) to calculate the time constants. Floru [5] and Abel [6] compared feedforward and feedback topologies in analog and digital domains respectively and proved that feedforward performs better compared to feedback structure. In [7], hardware implementation of a complete DRC algorithm is described in which the attack and release time constants are incorporated in the gain smoothing stage. The input level was estimated using a common time constant for both attack and release phases. The authors do not mention the effect of double stage filtering in the final output. In [8] [9] [10] also the time constants are estimated considering the input SPL only and the effect of gain stage is ignored.

In this paper, we examine the effect of compression ratio on time constants and propose a remedy for error occurring in the output of DRC. Further, the effect of smoothing stage on time constants is studied and a method to get accurate output that does not affect the hardware structure is proposed. The rest of the paper is organized as follows. In section II, basic DRC operation is explained. In section III, problems with conventional DRC algorithms and the proposed methods to overcome these problems are described. Simulation results and comparison with conventional methods are included in section IV and conclusions in section V.

II. DRC Basics

Dynamic Range Compressor modifies the amplitude of an incoming signal and gives an output with reduced SPL range. Block diagram of a basic feedforward DRC is shown in Fig.1. All DRC algorithms contain two fundamental blocks: Level detection and Gain stage. Level detector estimates the incoming signal level and converts it into logarithmic scale. The gain stage compares the estimated signal level with a predefined threshold value, Compression Threshold (CT) and calculates the compression gain according to a static gain curve as shown in Fig.2, in which L_{ti} and L_{to} represent input and output SPL corresponding to CT, and L_i and L_o represent instantaneous input and output SPL respectively. A smoothing stage is incorporated to reduce the effect of ripples generated due to Infinite Impulse Response(IIR) filtering and hardware implementation.

Static characteristics of a DRC include Compression Ratio (CR) and Compression Threshold (CT). CT decides the point

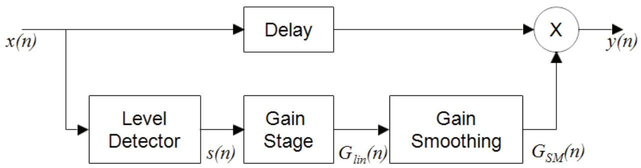

Fig. 1. Basic Block Diagram of DRC

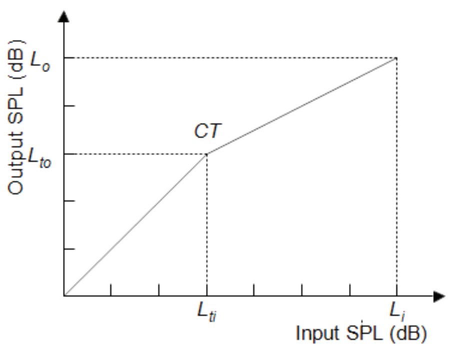

Fig. 2. Static Compression Curve

at which the gain changes in a DRC. CR is defined as the ratio of change in input SPL to the change in output SPL [11]. Compression Factor (CF) is the inverse of CR. The dynamic parameters, Attack Time and Release Time decide how fast the compressor should react to the change in input SPL. For the calculations we used ANSI S3.22 specifications for Hearing Aid characteristics. According to ANSI S3.22, Attack Time is the time taken by the compressor to reach within 3 dB of the final value of the output when the input changes from 55 dB to 90 dB SPL. Similarly Release Time is the time it takes to reach within 4 dB of the final value of the output when input SPL changes from 90 dB to 55 dB [11] [12].

III. Proposed Algorithm

Conventional approaches for DRC design and the modifications proposed to improve the algorithm are explained in this section.

A. Level Detection

Level detection stage estimates the input SPL as a moving average single pole IIR low pass filter. Two widely used methods include peak detection and RMS detection. In peak detection, the absolute or rectified amplitude of the incoming signal is used as input to the IIR filter. On the other hand in RMS detector, the squared value of the signal amplitude is given as input to the IIR filter. In this case, filter output is passed through a square root function to get the estimate of input SPL. Since the perceived signal loudness for a speech signal is considered to be closer to the RMS value of the signal amplitude, the latter method is preferred.

The input SPL is calculated using the following IIR equation in digital domain [9].

$$p(n) = \begin{cases} \alpha p(n-1) + (1-\alpha)|x(n)|, & \text{if } |x(n)| \geq p(n-1) \\ \beta p(n-1) + (1-\beta)|x(n)|, & \text{if } |x(n)| < p(n-1) \end{cases} \tag{1}$$

For RMS detector:

$$p(n) = \begin{cases} \alpha p(n-1) + (1-\alpha)x^2(n), & \text{if } x^2(n) \geq p(n-1) \\ \beta p(n-1) + (1-\beta)x^2(n), & \text{if } x^2(n) < p(n-1) \end{cases} \tag{2}$$

where α and β represent the attack time and release time constants respectively. $x(n)$ is the input signal amplitude at n^{th} sample and $p(n)$ represents the corresponding measured input SPL in linear scale. The attack time and release time constants are calculated by equating the unit step response of the above filter to -3 dB and -4 dB respectively [9] [7].

$$1 - \alpha^{N+1} = 0.7079 \tag{3}$$

where N is given by the product of required time constant and sampling frequency. For example, a sampling frequency of 20 kHz and attack time of 4 ms will give an attack time constant of 0.985. The estimated input SPL is converted to log domain for gain calculation.

$$s(n) = 20\log_{10}(p(n)) \tag{4}$$

In case of RMS detector the multiplying factor is 10 instead of 20 which will act as square root operation.

$$s(n) = 10\log_{10}(p(n)) \tag{5}$$

If the same attack time constant is used for RMS detector, the input SPL would reach faster to the 3 dB value. Hence (3) has to be equated to the square of 0.7079 to get an accurate time constant value for RMS detector. A modified equation for calculating the time constants for RMS detector is given below:

$$1 - \alpha^{N+1} = 0.7079^2 \tag{6}$$

B. Gain Calculation

Gain stage determines the amount of compression to be applied to the incoming signal according to the compression curve obtained from a particular fitting algorithm of hearing aid. The gain is computed in log domain using the following linear equation derived from the compression curve [6].

$$G_{dB} = (CR - 1)(s(n) - CT) \tag{7}$$

$$G_{lin} = 10^{(G_{dB}/20)} \tag{8}$$

where CR is the Compression Ratio, CT is the Compression Threshold and G_{dB} and G_{lin} are the calculated gain in dB and linear domain respectively. The gain in dB is converted back to linear domain and multiplied with the input signal to get the final output, y(n).

$$y(n) = G_{lin}x(n) \tag{9}$$

In (7) the expected value of gain corresponding to 3 dB value of output is affected by the multiplication factor, CR.

This multiplication factor reduces the expected gain by an amount of $3*CR$. This error in turn affects the rate of settling of output towards its steady state. By adding this extra factor in the gain equation the error can be removed. The modified equation for gain calculation is given as:

$$G_{dB} = (CR - 1)(s(n) - CT) + M \quad (10)$$

where M represents the multiplication factor which is equal to $3*CR$. This modification increases the computational complexity in hardware by introducing an extra multiplication and an addition term, which is not desirable. Hence, a modification is proposed in the calculation of time constants to cancel out the effect of unwanted multiplication factor M. Since the estimation of time constants are not carried out in real time, it will not affect the hardware structure. To get the modified time constant, a lookup table is formed by measuring the error in estimated gain and corresponding offset in input SPL for various values of CR ranging from 0.05 to 0.8 in steps of 0.05 through MATLAB® simulations for a test signal as given in Table I. A curve is plotted (Fig.3) between CR and the offset (δ) occur in input SPL. Then a quadratic equation of order 3 is derived using polynomial regression method, which will give the offset in input SPL corresponding to a particular CR, as given below:

$$\delta = 52.953CR^3 + 41.034CR^2 + 14.258CR - 0.6897 \quad (11)$$

New time constant is calculated by equating the unit step response in (3) to the modified value instead of -3 dB, which can be obtained by subtracting δ from -3 dB, in the case of attack time.

C. Smoothing

The filtering operation generates unwanted ripples. Similarly the various hardware implementation stages cause quantization errors. These errors induce distortion in the output signal which needs to be reduced. Smoothing stage helps in reducing these small artifacts in the output by blocking sudden changes in the output signal [4]. Smoothing is performed in the linear domain after converting the calculated gain value from log domain to linear. Similar to level detection stage a single pole IIR filter is used and the current gain value is updated till the difference between gain in previous sample and the updated gain becomes negligible. A low threshold value can be used to get better smoothing. The basic smoothing equation is given as:

$$G_{SM}(n) = \begin{cases} \alpha_{SM} G_{SM}(n-1) + (1-\alpha_{SM}) G_{lin}(n), \\ \qquad \text{if } G_{SM}(n-1) - G_{lin}(n) \geq G_{th} \\ \beta_{SM} G_{SM}(n-1) + (1-\beta_{SM}) G_{lin}(n), \\ \qquad \text{if } G_{SM}(n-1) - G_{lin}(n) < G_{th} \end{cases} \quad (12)$$

where α_{SM} and β_{SM} represents the attack time and release time constants for smoothing stage. G_{SM} is the gain at the output of smoothing filter.

In [7], a very fast time constant (corresponding to 4 ms) is used to estimate the input SPL which is common for both attack and release phases. The attack and release time constants are incorporated in smoothing stage. Since the filtering occurs twice (in level detection and smoothing), the effective time constants will be a combination of two stages in compressors which include smoothing stage. From simulations, it is found that initial time constant in level detection affects the later part and the resultant attack and release times were different from expected value. If identical time constants are used in both level detection and smoothing stages, a linear relation can be obtained between single filter time constant (expected value) and effective combination of two time constants in different stages in the case of peak detector based DRC. So a modified N for DRC with smoothing stage, which should be used in (3) while calculating time constant for both level detection and smoothing stages to get an accurate expected output is given as:

$$N_{SM} + 1 = 0.537(N + 1) \quad (13)$$

In RMS detector based DRC algorithms, the level estimation time constant is calculated by (6), but similar squaring effect is not present in gain smoothing stage. So if (3) is used to find time constant in gain smoothing and (6) is used for level detection stage a linear relation can be obtained between expected output and actual output. Hence, a modified N for estimating both the time constants can be obtained as:

$$N_{SM} + 1 = 0.424(N + 1) \quad (14)$$

IV. Results and Discussion

A step signal was chosen according to the specifications in ANSI S3.22 to test the algorithm, which characterizes a sudden rise from 55 dB to 90 dB after 600 samples, stays at 90 dB for next 1200 samples and a sudden drop from 90 dB to 55 dB at 1800^{th} sample. A sampling frequency of 20 kHz was assumed and an attack time of 4 ms was chosen which is lesser than short term stationary period of speech and longer than the fundamental period of sound waves [7]. CT was chosen as 70 dB. All simulations are carried out in MATLAB® 2015 with signal processing toolbox.

Table I is a lookup table formed by measuring the error in output SPL for different compression ratios. Actual sample value for each expected Final Value (FV) of output was measured from simulations and corresponding input SPL in

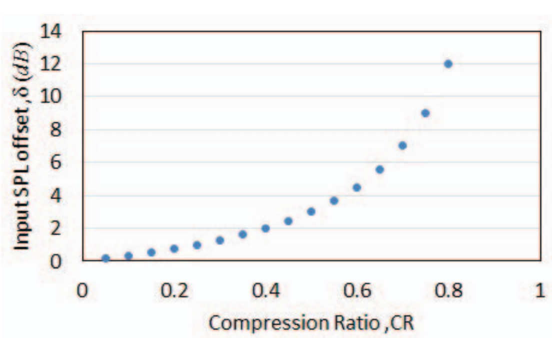

Fig. 3. CR v/s δ offset curve

TABLE I
LOOK UP TABLE USED TO DERIVE δ V/S CR RELATION

CR	CR*3	Final Value	FV+ 3 dB	linear SPL	Sample no.	i/p SPL	δ (dB)
0.05	0.15	71	74	5012	677	86.84	0.16
0.1	0.3	72	75	5623	674	86.66	0.34
0.15	0.45	73	76	6310	671	86.47	0.53
0.2	0.6	74	77	7079	668	86.25	0.75
0.25	0.75	75	78	7943	665	86	1
0.3	0.9	76	79	8913	661	85.72	1.28
0.35	1.05	77	80	10000	657	85.38	1.62
0.4	1.2	78	81	11220	653	85	2
0.45	1.35	79	82	12589	649	84.55	2.45
0.5	**1.5**	**80**	**83**	**14125**	**645**	**84**	**3**
0.55	1.65	81	84	15849	640	83.33	3.67
0.6	1.8	82	85	17783	635	82.5	4.5
0.65	1.95	83	86	19953	630	81.43	5.57
0.7	2.1	84	87	22387	624	80	7
0.75	2.25	85	88	25119	618	78	9
0.8	2.4	86	89	28184	612	75	12

linear domain was obtained. The offset (δ) was obtained by subtracting the measured input SPL from the theoretical value of 87 dB, i.e., (90-3) dB. For example, for CR=0.5, the output reaches the expected value of 83 dB at 645^{th} sample and it happens when the input SPL reaches 84 dB which is 3 dB less than the theoretical value of 87 dB.

Fig.4 and Fig.5 show the output of DRC with basic level detector for the step input with attack time constant calculated with 4 different methods: 1) conventional without smoothing, 2) proposed method without smoothing, 3) conventional with smoothing (level detection time constant kept as 4 ms) and 4) proposed with smoothing. CR is chosen as 0.5. According to ANSI specifications, during attack, the output should reach within 3 dB of the final value (80 dB), i.e., 83 dB in the 681^{st} sample. But in the case of conventional DRC without smoothing reaches the expected output at 645^{th} sample. In the case of conventional smoothing, the output reaches 83 dB at 750^{th} sample. Whereas for both the proposed methods the output reaches 83 dB at 681^{st} sample. A similar analysis is done for RMS detector as well and the results are given in Table II and Table III for peak detector and RMS detector based algorithms respectively. The results show that the proposed methods performed perfectly in all the cases.

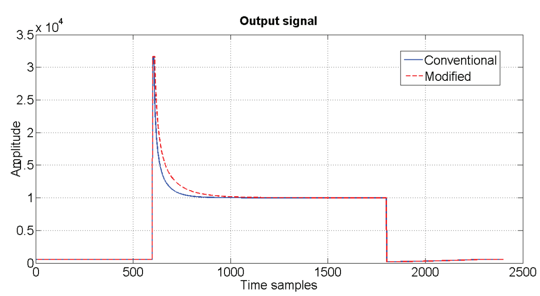

Fig. 4. Peak detector based DRC output without smoothing

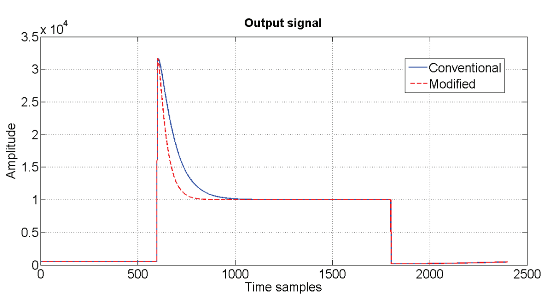

Fig. 5. Peak detector based DRC output with smoothing

TABLE II
SIMULATION RESULTS FOR PEAK DETECTOR

Parameters	Without smoothing		With smoothing	
	Conven-tional	Modified	Conven-tional	Modified
Attack time constant	0.985	0.9914	0.985	0.972
Output at 681^{st} sample(dB)	81.47	83	86.06	83
Sample value at 83 dB	645	681	750	681
Error(dB)	-1.53	0	3.06	0

TABLE III
SIMULATION RESULTS FOR RMS DETECTOR

Parameters	Without smoothing		With smoothing	
	Conven-tional	Modified	Conven-tional	Modified
Attack time constant	0.985	0.9964	0.985	0.9918& 0.9654
Output at 681^{st} sample(dB)	80.75	83	85.16	83
Sample value at 83 dB	619	681	730	681
Error(dB)	-2.25	0	2.16	0

V. CONCLUSION

This paper presents methods to estimate the Attack Time constant for Dynamic Range Compressor algorithm in Hearing Aids. Conventional methods calculate the time constants without considering the problems imposed by the gain and smoothing stages in DRC. The equations are modified to calculate these parameters in such a way that a perfect output gets generated as specified in the ANSI S3.22 standard for hearing aids. The simulation results show that the proposed algorithms work as expected for both peak and RMS level detector based DRC algorithms with and without gain smoothing stage.

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