Analysis of an alternative approach to digital domain volume control claiming high perceptual audio quality

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ABSTRACT

In an increasingly digital environment, digital domain volume control offers a cost-effective alternative to analogue domain implementations. However, it has not found widespread use in the upper segment of the consumer audio market. Even if advances in digital to analogue converters and signal processing technologies have addressed the shortcomings of early days' implementations, many audiophiles still complain about inferior sound quality of digital solutions. In this Engineering Brief, we analyse an alternative approach to digital domain volume control said to have scored high in casual subjective listening tests. Looking at bit-level arithmetic and information propagation considerations, we present objective elements which distinguish the new volume control from traditional approaches. These may explain the reported superior perceived audio quality.

1 Introduction

Digital domain volume control offers an alternative to analogue domain solutions, supported by the promises of simplified system design and cost benefits. Whilst early days' implementations suffered from the limited resolution of available digital to analogue converters and digital signal processors, technology has now matured sufficiently to allow for practical implementations. Progress in signal processing algorithms has allowed to overcome the problems linked to quantization noise and modern converter offer sufficient dynamic range. Still, digital domain volume control has not yet found wide application in the upper segment of the consumer market, with audiophiles complaining about inferior quality of digital domain solutions.

In this paper, we analyse a new type of digital volume control, claimed to have superior audio quality. In section 2 we review the existing state-of-the-art and in section 3 we present the principles of the alternative approach. Section 4 analyses the differences between digital volume control solutions. Section 5 proposes a new combined approach and section 6 provides the conclusions.

2 Overview of digital domain volume control

Digital domain volume control on PCM audio data is simply implemented by multiplying incoming audio samples by the desired volume control coefficient. More precisely, let's consider that the input samples are M bits signed integers in two's complement representation, MSB aligned in N bit words (with $N >= M$, N being the system word width) and that the volume control coefficient is a K bits unsigned integer. Without loss of generality, we can consider that volume control is limited to attenuation. In this case, the result will be a $N+K$ bits signed integer and extracting the N most significant bits will provide the desired result as shown on Figure 1. Clearly, usability of such a digital volume control is highly dependent on the number of bits N used to represent the audio samples.

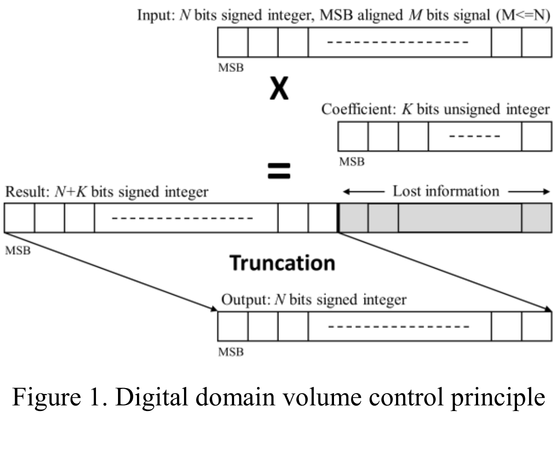


Figure 1. Digital domain volume control principle

For instance, applying a 30dB attenuation on 16 bits audio samples ($N=16$) would leave an output dynamic range of only 11 bits, which is insufficient. This shows why digital volume control was not a viable solution in the early days of digital audio.

Quantization noise resulting from the truncation of the N+K bits result to N bits was another issue with early days digital volume controls. This quantization noise is correlated with the input signal as shown on Figure 2, which is detrimental to perceived audio quality. It has been shown (Ref [1] to [4]) that addition of triangular probability density function (TPDF) dither spanning +/- 1 least significant bits (LSB) before truncation allows for decorrelation of the signal and quantization noise. This is illustrated on the right hand side of Figure 2.

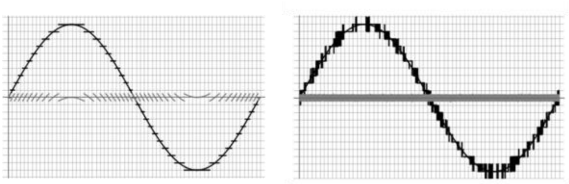


Figure 2. Quantization noise and TPDF dither

However, despite quantization noise decorrelation and increased audio samples word width (N = 24 or more) which address the apparent concerns of early implementations, many audiophiles still report quality loss when using digital domain volume controls compared to analogue solutions.

3 Alternative approach to digital domain volume control

The alternative approach to digital domain volume control (named Leedh Processing Volume), has been proposed by Mr. Milot from French company Acoustical Beauty (Ref [5]). Patent protection at international level (Ref [6]) has been applied for. The key idea behind the alternative approach is to minimize information loss at truncation stage. For instance, consider a signal represented in 24bits words (MSB aligned). If volume control coefficients were limited to 8 bits and the input signal had 16bits resolution, no truncation would occur and no information would be lost. This trivial example hints towards the principle governing the alternative digital domain volume control: Minimise the number of bits K used to quantize volume control coefficients so that information loss is minimized at truncation stage. In other words, **it trades volume control coefficients precision against information loss minimization**.

As an illustration, consider a volume control using 1dB steps. Table 1 shows the volume control coefficients used by the alternative approach for the top 6dB range (-1dB to -6dB). The full coefficients table is then built by shifting these values by 1 bit to the right (i.e. increasing K by 1) for each 6dB slice.

Nominal attenuation	K	Coefficient value	Effective attenuation
-1 dB	3	7	-1.16 dB
-2 dB	4	13	-1.80 dB
-3 dB	4	11	-3.25 dB
-4 dB	3	5	-4.08 dB
-5 dB	4	9	-5.00 dB
-6 dB	1	1	-6.02 dB

Table 1. Top 6dB slice in 1dB step.

As can be seen in Table 1, K is drastically reduced compared to standard coefficients choices which would typically use values for K of 16 or more. The price to pay for this reduction is the reduced precision in attenuation value. One can observe that for N = 24, the example allows for truncation free volume control of 16 bits signal up to about -30dB. Acoustical Beauty has reported that the alternative digital domain volume control has scored high among audiophiles in casual listening tests (against both digital and analogue volume controls). The next section examines if there are objective elements that could corroborate these findings.

4 Analysis

In this analysis, we examine frequency domain behaviour and information propagation at bit arithmetic level. We consider N = 24, K = 16 and M = 16 (i.e. input signal is provided as 16 bits data MSB aligned in 24 bits words and volume control coefficients are 16 bits wide) as an example but results can be extrapolated to other cases. The alternative volume control coefficients are selected according to the example described in Section 3.

Frequency domain analysis

We consider an input signal consisting in an un-dithered 16 bits, 0dBFS, 1.5kHz sine wave sampled at 48kHz. This results in a repeating sequence of 32 samples and a power of 2 length FFT clearly shows the harmonics due to 16 bits truncation as shown in Figure 3.

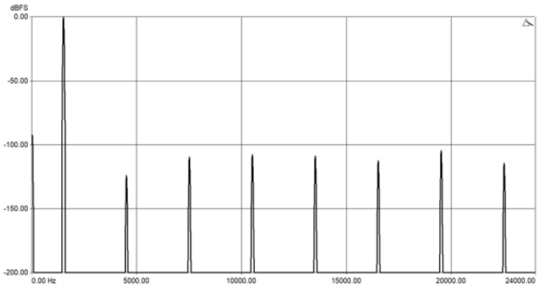
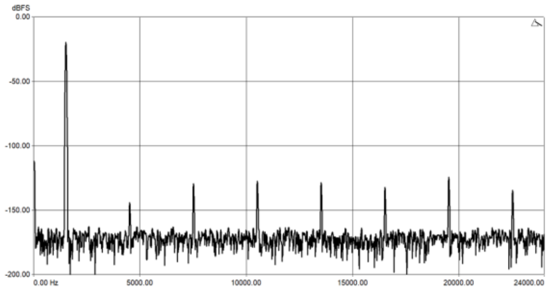


Figure 3: spectrum of 16bits, 0dBFs, 1.5kHz sine wave sampled at 48kHz

Figure 4 illustrates the spectrum of the same signal when different types of volume control are applied for 20dB of attenuation. It clearly shows how the alternative volume control achieves distortion free volume control. The effective attenuation is -19.8dB and is expressed on *K'* = 7 bits. The resulting spectrum is clearly identical to the one of the original input signal as we have $M + K' <= N$.



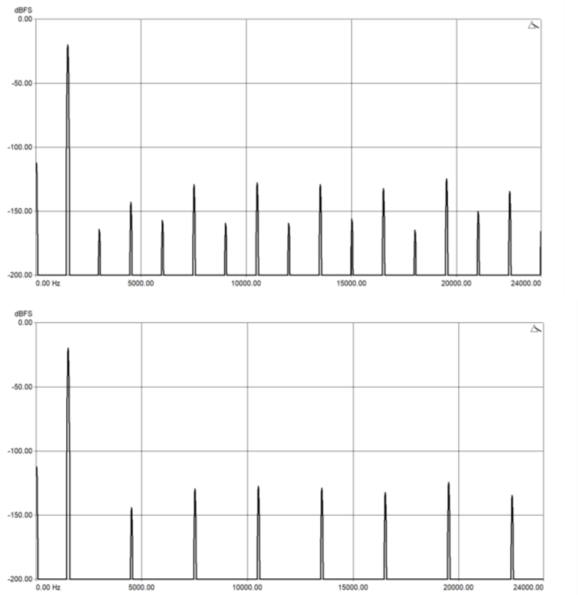


Figure 4: spectrum of 16bits, 0dBFs, 1.5kHz sine wave sampled at 48kHz, attenuated by 20dB using:
Top: standard, un-dithered volume control
Middle: alternative volume control
Bottom: standard, dithered volume control

Figure 4 also shows how an un-dithered standard volume control introduces truncation distortion and how dithering cancels unwanted harmonics at the price of added broadband noise.

Info propagation analysis

We now analyse how signal information is propagated through the volume control. We propose to visualize the multiplication process of an *N* bits integer by a *K* bits unsigned integer in a 2D representation as shown on Figure 5. The grey zone on the left represents truncation of the output to *N* bits.

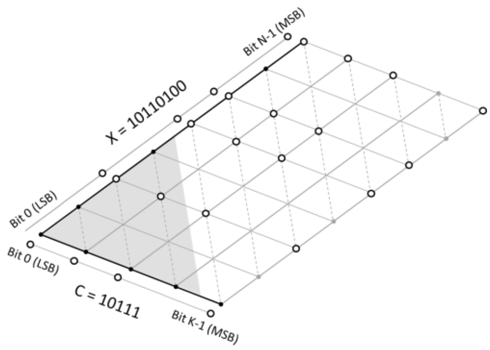


Figure 5: 2D representation of *N* x *K* bits multiplication (N = 8, K = 5)

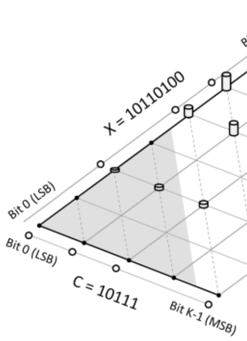
Let's consider an input signal x and a volume control coefficient c defined by

$$x = \sum_{i=0 \ldots N-1} 2^i x_i \qquad c = \sum_{j=0 \ldots K-1} 2^j c_j \qquad (1)$$

The product of x by c is made up by summing all terms of the form $2^{i+j} x_i c_j$ such that $x_i c_j \neq 0$. For each of these terms, we define the Signal Information Relative Contribution (*SIRC*), where bits are inverted for negative signals, as:

$$SIRC(i, j) = \frac{(i+1)}{N} \times \frac{(i+j+1)}{N+K} \qquad (2)$$

This defines the signal information contribution for a basic product term. If we now plot the *SIRC* at each basic product term node of the 2D representation of N x K bits multiplication introduced above, we get the Signal Information Propagation Map (S*IPM*), of the product $x \times c$. Figure 6 shows the *SIPM* for a *N* x *K* bits multiplication (*N* = 8, *K* = 5), once with c spanning the full *5* bits to illustrate a standard volume control and once with c spanning only 2 bits to illustrate the alternative volume control. As can be seen, the traditional volume control suffers from possible information loss as some *SIPM* elements fall below the quantization level of the output signal.



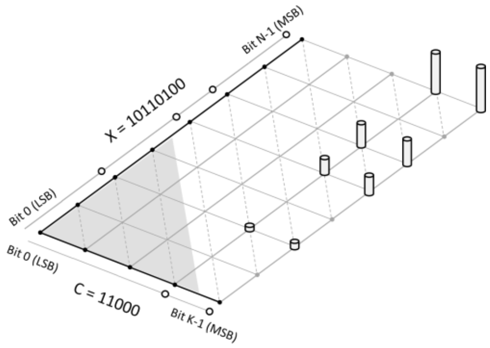


Figure 6: *SIPM* for traditional (Top) and alternative (Bottom) volume controls

In addition, as lowering the volume roughly corresponds to shifting the *SIPM* along the coefficient axis towards the LSB, we can observe that the alternative volume control allows for higher attenuation values before significant signal information gets lost below the output quantization level. We can now define the Signal Information Propagation Index (*SIPI*) for an input sample x and a volume control coefficient c as the average of *SIRC* values across the corresponding *SIPM*:

$$SIPI(x, c) = \frac{1}{P} \sum_{x_i c_j \neq 0} SIRC(i, j) \qquad (3)$$

where *P* is the number of *(i,j)* for which $x_i c_j \neq 0$.

Finally, for a volume control consisting in a set C of volume control coefficients, we define the Signal Information Propagation Score (*SIPS*) as the average of *SIPI* across all possible input signal samples and volume control coefficients:

$$SIPS(C) = \frac{1}{Q} \sum_{x,c} SIPI(x, c) \qquad (4)$$

where Q is the number of (x, c) couples. Similarly we define *SIPS'* as being equal to *SIPS* except that *SIRC(i,j)* is replaced by 0 if *i*+*j* < *K*. *SIPS'* removes information contributions that are below the output quantization level. *SIPS* and *SIPS'* provide scalar measures of signal information propagation. Higher values mean better signal information preservation. Table 2 shows *SIPS* and *SIPS'* for the standard and the alternative volume controls.

	Standard	Alternative
SIPS	0.37	0.42
SIPS'	0.16	0.27

Table 2. *SIPS* and *SIPS'*

The alternative volume control shows advantages in terms of *SIPS* and *SIPS'*. However, as correlations between these scores and subjective listening tests have not yet been studied, we cannot conclude that this is the reason for the alternative approach's reported superior subjective audio quality.

5 Combined approach

As exposed in Section 3, the key principle of the alternative volume control is to trade coefficient precision against minimized truncation noise. Nevertheless, truncation can still happen with the alternative approach. Let's consider the example analysed in Section 4 but with N = 20. We observe that truncation noise is now present at the output as shown on Figure 7. As an improvement, we therefor propose to add a TPDF dithering stage at the N^{th} bit level before final truncation. Apart from eliminating quantization noise related distortion, this would provide a constant noise floor, which may be of benefit in terms of perceived audio quality.

6 Conclusions

The analysis presented in Section 4 has shown that the alternative approach provides distortion-free volume control (up to a certain level of attenuation) and apparent advantages in terms of information propagation.

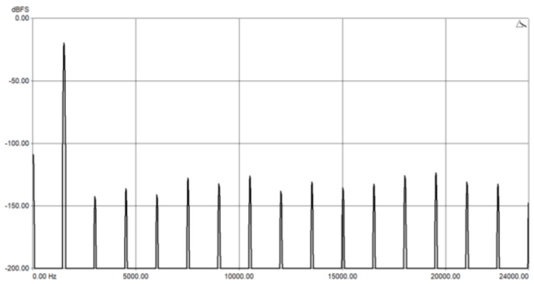


Figure 7: spectrum of 16bits, 0dBFs, 1.5kHz sine wave sampled at 48kHz, attenuated by 20dB using the alternative approach and quantized to 20 bits

More precisely, if we consider that signal information preservation prevails over exact target attenuation value, the alternative seems at advantage. However, we cannot conclude that this advantage correlates with superior subjective audio quality as no data resulting from a significant study is available. Finally, in Section 5, we have proposed to add TPDF dithering to the alternative volume control to avoid truncation distortion that may appear at higher attenuation settings.

References

[1] J.R. Stuart, P.G. Craven, "The Gentle Art of Dithering", *JAES,* vol. 68, no. 5, pp. 278—299 (2019)

[2] S.P. Lipshitz, J. Vanderkooy, "Dither Myths and Facts", *117th AES Convention, San Francisco,* Paper 6279 (2004)

[3] S.P. Lipshitz et al, "Quantization and Dither: A Theoretical Survey", *JAES,* vol 40, no. 5, pp. 355-375 (1992)

[4] R.A. Wannamaker, "The Theory of Dithered Quantization", *University of Waterloo,* PhD Thesis (2003)

[5] www.leedh-processing.com
www.processing-leedh.com
gilles.milot@acoustical-beauty.com

[6] Patent application WO2019076783A1