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DESC 9115 Written Review 2

Multi-Band Limiter Value to Production Chain and Its Processing Limitation

Nowadays, digital signal processing in audio production environment has been synonymous and highly incorporated by most audio engineers. The main reason for it is it's cost efficient to implement and as we advance in digital signal processing its able to substitute our analogue rack with a simple program that we can just dial in. Digital signal processing has revolutionize the audio production industry as it is less messy in terms of setup and the digital side of it, kept things simple to work with while it enables engineers to experiment with less fuzz. As technology vastly improves and the cost to have one has been significantly competitive, digital signal processing has already been a blessing one might say while there are consequences of its usage on our production chain that needed to be considered. Within this review we will discuss multi-band limiter as an audio production tool; why do you want it and limitations generated by multi-band limiter.

1. Introduction

A limiter purpose just like its name is to limit and prevent dynamic range of a signal input from clipping which is undesirable in audio production process. When the input is below predetermined level widely called threshold, the audio dynamic is not change at all and the entire signal above that threshold will be limited amplitude wise [1]. Limiters are often use in broadcasting and CD mastering to protect the signal chain from clipping which tend to produce distortion when the level goes unlimited. Limiter also found in a lot of active transducer or power amplifier to protect our devices from burning out caused by the excessive amount of amplitude that the device able to handle. Some devices even have it design to automatically idle respective device into a steady state to provide an overload protection that might destruct the circuit board or blow a transducer.

In the contrary to analogue device, digital system does not provide a warning and it suddenly clips out when 0 dBFS is reached. Moreover it lets us to push up our audio level while maintaining the same peak level. Limiter has been an integral chain to produce loudness within the industry particularly in broadcasting and audio mastering since producers want their mix to stand out in comparison to others. Television station are the obvious one and we can hear it where it became very loud during commercial as they want to suck up people's attention to the particular part of the program.

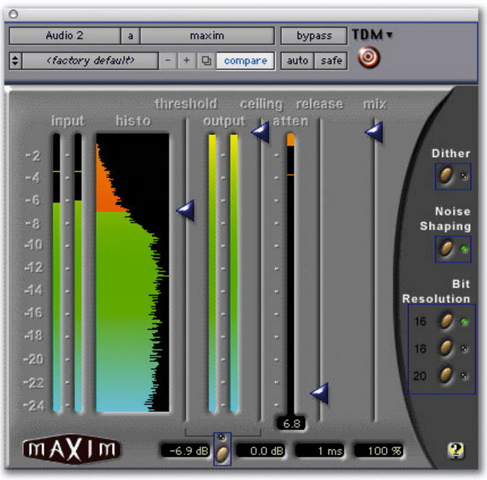

Figure 1. Digidesign Maxim Limiter

There are two types of limiter, which are single-band and multi-band limiter. Again, as it is called multi-band allow us to control the threshold value at each individual frequency band done through band frequency splitting process, limiting and the recombination of each band output into a single output while single-band allows the control of the full bandwidth provided by the input signal. In practice, using a single band limiter might be troublesome during mastering process due to a bad mix usually influenced by room acoustics or speaker in used to mix them. For examples, doing full bandwidth level adjustment can results in over dominance of particular frequency that makes it clip earlier than we expect them to hence reducing the headroom to work with.

In the opposite individual threshold control provides engineer more headroom to work with while in the industry it is widely known that executive producers wants loudness to feature in what they are selling with actually does some damage to markets perception in loudness. Multi-band limiter reduces "pumping" effect during mastering process and in broadcasting it help to limit unnecessary frequency band at the consumer end due to their playback limited frequency such as TV set or standard car audio system. In broadcasting it also helps to transmit the audio in a lighter fashion therefore consumer can have a better clarity over the end products.

It is also useful to work as a dynamic equalizer which conventional equalizer only work well to reduce resonances at specific frequency. In practical application, engineers able to boost a particular band without taking so much headroom since we can set a predetermined level to prevent it from over-dominating. Moreover, we can also increase the warmth of a track by increasing the level of low mid while equalizing it would make it sound "muddy" [1].

Figure 2. Waves L3 Multi-Maximizer

2. Signal Flow Processing

Band splitting process is the key to any multi-band signal processing. It is done through the implementation of several key filter applied to the input signal that will essentially divide the signal into several band for its own purposes. Figure 2 illustrates how 3-band limiter processing signal flow is done; nevertheless applying more filters to the processing chain can increase the numbers of band involve within. Generally, the filters in use are Infinite Impulse Response (IIR) filter and Finite Impulse Response (FIR) filter that utilize the impulse response (IR) of the signal to generate the filter intended for the filtering purposes.

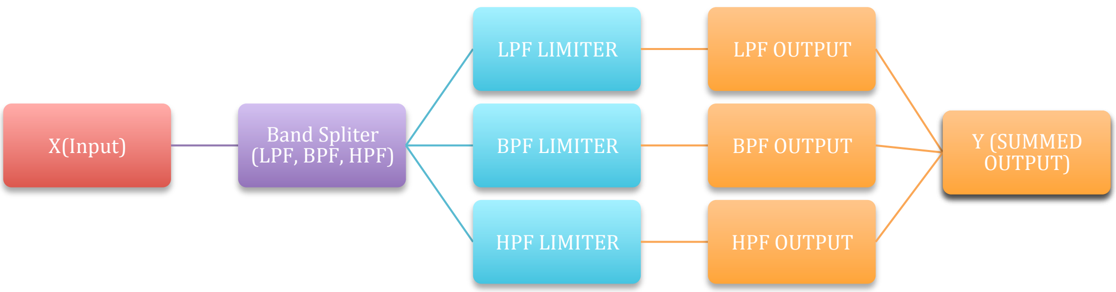

Figure 3. Signal Processing Diagram

IIR filter is highly effective to work in wider bandwidth that does not require extreme resolution. It is widely used in active sub-woofer processing to filter out high frequency band to have better transducer headroom. However it does not explicitly filter the all of higher frequency content and that's why we can hear higher frequency content coming out of our subwoofer as the results of a full bandwidth playback. Due to the nature of IIR filter that provides feedback into the system (see figure 4), the resolutions of IIR filter cannot be too high otherwise it would not stop calculating which might cause never-ending processing that we simply not want it to happen.

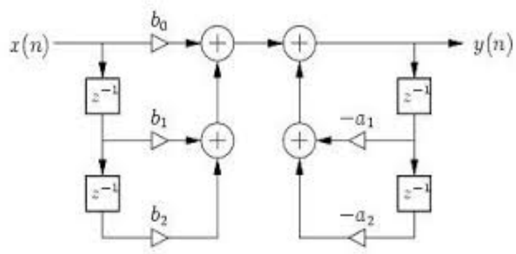

Figure 4. IIR System Design

IIR Filter is effective in creating peaks at the intended frequency with lower bandwidth resolution while it is also sufficient in creating wide notches above the crossover frequency. On the other side FIR filter is able to overcome the problems given by IIR filter. FIR main advantage is, its capability to maintain a linear phase characteristics. Linear phase refers to the condition where the phase response of the filter is a linear function of frequency.

Since the filters have constant group delay, all frequencies will results in the delay through the filter being equal at all frequencies while the number of coefficient in use determines the performance of FIR filter, the increase of. It is utilized because it can produce a strong reduction on unwanted frequencies while also eliminating the risk of infinite calculation caused by the feedback; moreover it also has its shortcoming that could be problematic, particularly addressing its processing requirement.

3. Shortcoming of FIR Filter

Digital audio processing requires a massive amount of processing power that also dependent of the intended outcomes; for instance to create 2 minutes of an auditory scene for an animated motion pictures some engineer might require to have more than 200 tracks to produce the desired outcome. Therefore you can imagine that processing efficiency is an integral part to the production chain otherwise we will experience lagging or inaccurate reproduction due to inadequate processing power which is going to slow down our workflow and definitely produce annoyance.

In spite of the accuracy FIR Filter able to generate and its effectiveness in creating a filter, for it to run swiftly it needs a significant amount of processing power due to the higher degree of filter resolution that we want to produce. Another undesirable products of FIR is it behaviors in maintaining ringing artifact due to the stored energy that must be released and generally any cut-off filter would produce ringing artifact regardless of the input signal.

The fact is, the energy is effectively stored within the filter and then releases by the filter as the temporal factor progress that led to ringing artifacts. Often this ringing artifact goes unnoticed because it is buried by the response to the in-band signal that presents. However, if the in-band signals that present were stopped out of sudden due to the production arrangement this ringing would be audible and it sounds quiet similar to a reverberant tail because FIR filter will continue to empty the energy stored even after the signal ends.

4. FIR Filter Coefficient Optimization With Complexity Aware Algorithm

Since FIR Filter needs a hefty amount of processing power there are a way to reduce the processing requirements up to 51% by applying coefficient optimization. Most DSP kernels are transforms with fixed coefficient where the area can be further reduced by common sub-expression elimination (CSE) while adders and shifter can replace the constant multipliers; however hardware complexity is not taken into control during the quantization process.

Using iterative algorithm to distribute the signed power of two terms (SPT) proposed by Li et al [3], will provide an estimation to the added quantize coefficient although it will lead to less optimal design especially when CSE is applied [4]. Complexity aware algorithm is quantizing the coefficient by allocating SPT terms under a precise adder budget while also taking into account the CSE where the combinations can reduce up to 51% of processing requirement.

Complexity aware algorithm controls the CSE heuristic to ensure the minimum allocated addition during successive approximation. Particular QC's is initially started at zero and then an SPT term is continuously assigned to the QC [4]. Once the allocated SPT remain stable, CSE is performed to introduce new stability. Therefore a feed-forwarded is necessary to insert more significant SPT term. Once the term is added, the less important zero overhead SPT terms should be removed. Last but not least additional CSE is performed to check if there are any new order for an improve CSE.

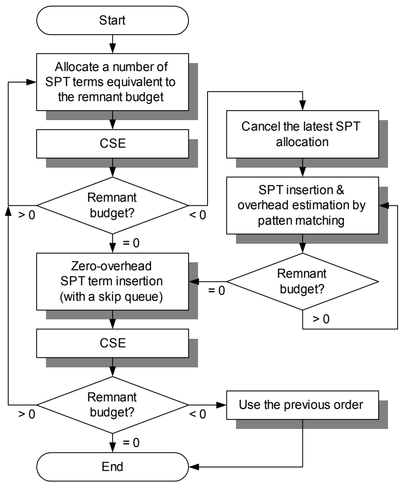

Figure 5 [4]. Complexity Aware Algorithm Scheme

Ultimately, Complexity Aware Algorithm will quantize the coefficients of FIR filter, which will accurately pre-defined addition stability to the quantized coefficient by eliminating the common sub-expression. The algorithm succeeds to apply an optimal scale factor within the gain tolerance to collectively stabilize into the quantization space [4].

5. Conclusion

To sum up multi-band limiter is a very useful production tools to be incorporated onto audio production chain. Digital audio processing is viewed to be an excellent future of the industry while not neglecting the proven quality of an analogue device both can be combined to achieve an optimum result. In a world where technology improvement happens everyday digital audio processing is proven to overcome production inefficiency such as bulky hard case on live production, faster to set up, less wiring problem. Technology improvement will bring down the cost of processing therefore it is viewed as an opportunity to grow and emulate the performance provided by analogue devices. Even though digital signal processing hasn't reached the height yet, it is realistic to think that more improvement will occur due to the opportunity and the possibility to improve existing methods and technology.

6. REFFERENCE

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