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An automatic maximum gain normalization technique with applications to audio mixing.

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ABSTRACT

A method for real-time magnitude gain normalization of a changing linear system has been developed and tested with a parametric filter design. The method is useful in situations where the maximum gain before feedback is needed. The method automatically calculates the appropriate gain that should be applied in order to maintain maximum unitary gain. The method uses an impulse measurement of a mathematical model of the system to be normalized. This is particularly useful for mixing engineers, who have to continually revise their gain structure in order to maximize gain before feedback. The system is also useful in many other situations where solving the analytical solution from the mathematical model is not possible.

1. INTRODUCTION

Public addressing systems that use a microphone amplifier speaker chain to transmit sound through the air towards the listener are essentially a feedback system. Using the air as a propagation medium has the inevitable effect of turning the sound reinforcement system into an endless feedback loop, and it is the air itself that acts as a feedback path. This is an inherent property of a sound system, and must be taken into consideration when designing or interacting with the system. The design aim is to reduce audio artifacts due to the feedback path. With this goal in mind, it is the purpose of this paper to introduce a normalization technique that prevents feedback when interacting with an audio system. The proposed method automates the engineering task of continually revising the system gain structure in order to avoid undesired feedback artifacts. This method permits one to achieve maximum gain before feedback while realizing the technical constraints of the mixing engineer, thus permitting him to concentrate more on the aesthetic contributions of the mixture. The method permits the audio mixing engineer to interact with the system without the fear of introducing feedback. The algorithm uses an impulse measurement of a mathematical model of the system to automatically calculate the appropriate gain compensation to avoid undesired artifacts due to feedback.

1.1. Understanding acoustic feedback

Feedback is the result of a retro-alimentation of the output signal of a system to its input. In an acoustic system these artifacts are introduced due to the feedback path and can be positive and negative feedback contributions. A simplified diagram of an acoustic feedback system is presented in Figure 1. The source signal is picked up by a microphone, transformed by equalization, amplified and played back through a speaker at the output. This is then attenuated and delayed as the output is transmitted through air, and summed with the input signal. $H_{ETOT}(x)$ is the electronic feed-forward transfer function of the system, and it is the result of the product of the individual transfer functions of the signal chain given by the microphone equalizer amplifier and speaker. $H_{ATOT}(x)$ is the acoustic transfer function of the system.

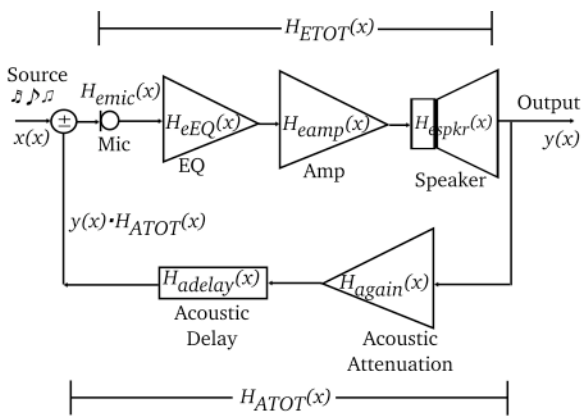

Figure 1 Model of a sound reinforcement feedback system.

For this paper we will only be concern with undesired feedback phenomena, properly known as howling [1] or Larsen effect [2]. This is a state in which system gain exponentially increments out of control, causing an undesired audible pitch. The feedback causes the audio system to behave in an unstable manner. Therefore, this condition must be avoided at all cost.

Given the acoustic model in Figure 1, the system will introduce undesired howling artifacts if equation 1 is satisfied.

$$H_{ETOT}(x) \cdot H_{ATOT}(x) >= 1 \qquad (1.)$$

If, for example, the equalizer transfer function gain, $H_{eEQ}(x)$, is 0dBs when flat and the overall electronic transfer function of the system $H_{ETOT}(x)$ is on the marginal condition before howling, then boosting the equalizer will introduce an undesired feedback artifact, and performing a cut on the equalizer will permit the system to remain stable. Therefore, a normalization technique which enables relative gain changes while forcing the transfer function of a linear system to have a maximum peak of 0dBs will preserve the stability of the system .

1.2. Achieving maximum gain before feedback.

To maximize the acoustic gain while avoiding feedback, the system should have a flat frequency response which falls below the threshold for acoustic feedback. Figure 2 shows the acoustic measurement of the frequency response of an audio system before and after optimization. The 0dB mark represents the threshold before feedback. The area between the frequency response and the 0dB mark represent unused system gain. It is the goal of an audio system engineer to minimize this unused area by flattening the frequency response of the system. This ensures a system with no coloration with the added benefit of maximizing gain before feedback. To achieve maximum gain before feedback audio operators have relayed mainly on equalizers [3], delay and feedback cancellation techniques.

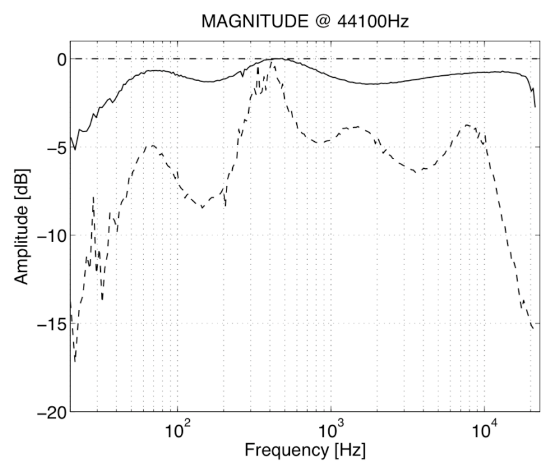

Figure 2 Acoustic measurement of the frequency response of a audio system. The dash-dotted (-·-·-) line represents the threshold for maximum gain before feedback, the dashed line (- - -) represents the frequency response of a non-optimised acoustic system and the full line (——) is the frequency response of an optimized quasi-flat system.

In recent years, our understanding of the acoustic feedback phenomena and when equalization can be achieved has improved. Currently, measurement techniques like, time delay spectrometry [4] and source independent measurements [5] have become more widely available, making the use of equalizers and delay lines more of a technique rather than a matter of skill. Also current design techniques and modern electronics, acoustics and speaker technology make a flatter frequency response a reality. Although the proper design of audio systems still requires a great amount of knowledge from the system engineer, a close to flat frequency response system which maximizes the gain before feedback is now a reality. The process of achieving this is commonly known in the industry as aligning, in time and frequency, a system. The full details of this process are beyond the scope of this paper but more on this can be found on [6].

The other important method of achieving maximum gain before feedback is by the use of feedback cancellation. Currently there are four main feedback-controlling techniques [2]. The first one consists of slightly frequency shifting the output signal so that the electronic transfer function is out of alignment with the acoustic transfer function, this causes a destructive interaction between the input and the acoustic feedback path, which effectively reduces feedback. In practice it can achieve up to 3 dBs increase in gain before feedback. This method is effective for speech applications but is not suitable for music. This is due to the simple reason that it modifies pitch, which would result in undesired atonal music.

The second feedback control technique is the all-pass filter approach. This is used to invert the phase of a potential feedback frequency. Unfortunately this technique is only useful with low delay systems with a prominent resonance. When applied to a system with flat frequency response it causes the feedback to jump endlessly from one section of the spectrum to other. For this reason its use is very limited.

Third is the adaptive filter modeling [7]. This uses technology based on echo-cancellation, aimed on telecommunication applications. The main idea is to subtract the far end speech from the near end speech. When the model is accurate it can achieve up to 10dBs of added gain before feedback. Due to the closed loop nature of the acoustic audio system the residual error of this process are highly correlated to the signals involved, and this can cause noticeable artifacts. When the model deviates it can introduce distortion and artifacts. It can even cause undesired feedback artifacts, which should not have been there. For this reason it has mainly been applied for speech systems where conditions are controlled. It is currently not consider a good candidate for sound reinforcement.

Finally, there is the adaptive notch filter method [8], which consists of a series of fixed and non-fixed notch filters, which filter out feedback frequencies when detected. The system performance is a trade-off between speed of detection and accuracy, and can notch out program material if a feedback discrimination system is not implemented properly or the system is overused. This method is highly effective and is widely used on sound reinforcement applications. Unfortunately it does not offer any extra gain before feedback for a flat frequency response system.

Currently, there is no optimal feedback cancellation method for music which offers a substantial improvement in gain before feedback without dangerous side effects. Therefore, it is the belief of the authors that if system alignment and an acoustic flat frequency response are currently achievable, then there is little need for feedback cancellation techniques. For this reason, we present a normalization technique which helps preserves system stability rather than another feedback cancellation technique. The aim is to prevent howling before it happened rather than suppress it after it has happened.

2. NORMALIZATION TECHNIQUES

Normalization of a signal consists in dividing the output by a given constant. In our case we are interested in normalizing the output signal of a linear system with the aim of keeping its overall maximum gain to be one, or 0 decibels full scale (dBfs). For this the normalization constant will correspond to the inverse of the maximum of the transfer function of the system under study. Such a normalization system has a power reduction proportional to the normalization constant. The goal of the methods presented in this and the following sections is to find the maximum of the transfer function in order to normalize the system. In this section, two normalization methods will be discussed; their advantages and disadvantages will be analyzed. In section 3, we will propose an alternative normalization technique.

Finding the maximum value of a transfer function composed of multiple elements, such as a parametric equalizer composed of multiple varying filters, is not a trivial task. Even if one knows the individual maxima of each component of the transfer function (such as through a parallel or series decomposition), their interaction can result in a maximum located at a completely different location. Given that the user can change the coefficients at any time to adjust the processing system, for example to modify an equalization filter, it becomes an even more challenging problem. In fact, the location and magnitude of the maximum of the transfer function is the result of the complex interaction of simpler transfer functions with each other. Therefore this involves both phase and amplitude interactions.

2.1. Mathematical Normalization Approach

Given that the coefficients of the transfer function can be changed by the user at all times, a familiar approach to finding the maximum, is finding the analytical solution of the roots of the first derivative of the transfer function. This approach requires a discrimination process in order to separate the local maxima from the global maximum. The steps for performing such an approach are presented next:

Given a Laplace domain transfer function:

1) Substitute terms so that the transfer function is in terms of the frequency.
2) Calculate the derivative with respect to the frequency.
3) Find the roots for the result obtained on step two.
4) Solve the roots and discard all results but the largest number.

Once the maximum has been found, the input is then divided by this maximum amplitude in order to maintain the system under unity gain. This method has the advantage that it can be implemented at clock speed rather than at sampling rate speed. It is highly effective for simple transfer functions, but unfortunately for most complicated cases, such as a transfer function representing a six filter parametric equalizer, it becomes practically impossible to find the exact analytical result for the roots. Thus, this approach is limited to static coefficients or to a more elaborate mathematical approximation. Such advanced mathematical approaches must be tailored to each particular case of linear system under study. In many cases, this means re-implementing the complete normalization design.

2.2. Real Time Transfer Function Measurement Normalization

A more general solution to the normalization problem is to measure the transfer function of a linear system such as the one depicted in Figure 3 using a source independent measurement algorithm. This approach has the advantage of working for all linear systems without the need of re-implementation for more complex systems.

$$x(t) \longrightarrow \boxed{H} \longrightarrow y(t)$$

Figure 3 Model of a linear system

The exact transfer function *H* of the system in Figure 3 is given by dividing the Laplace transform of the output by the Laplace transform of the input. In source independent measurement the transfer function is approximated by dividing the Fast Fourier Transform (FFT) of the output by the FFT of the input, equation 2. The approximation is due to the finite size of the FFT frame. Further improvements to this approximation are presented in [9].

$$H \sim= FFT(x(t)) / FFT(y(t)) \qquad (2.)$$

To use such measurement an algorithm implementation such as the one shown in Figure 4 is needed. In this implementation, the source independent measurement algorithm performs a continual reading of the input and the output and performs a division of its corresponding FFT frames synchronized in time. The result is post processed to improve accuracy and finally a maximum peak detector is used to determine the transfer function maximum. The inverse of this maximum value is then used to multiply the input in order to maintain the system under unity gain.

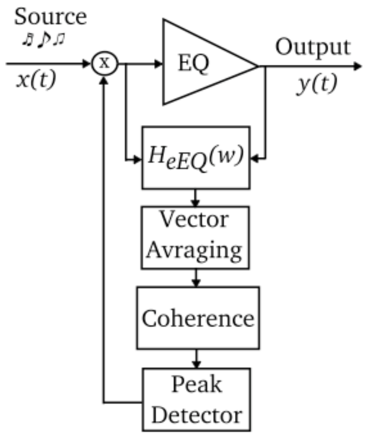

Figure 4 Real time transfer function normalization using source independent measurements.

Unfortunately this approach has to be implemented at a sample rate speed which makes the algorithm slower than a purely mathematical implementation. Also in order for this algorithm to give a precise measurement a number of frames must be averaged, and coherence and threshold techniques are required before calculating the maximum peak. All of this can be overcome, to some extent, by compromising precision and by algorithm optimization. Lack of precision will translate into a peak measurement which is non-stable and will cause the input to be modulated, introducing undesired audible artifacts. On the other hand a slow performance may cause the system level to go beyond 0dBfs for small periods of time which can introduce temporary undesired feedback artifacts.

3. PROPOSED AUTOMATIC MAXIMUM GAIN NORMALISATION TECHNIQUE

The main idea of this normalization technique is to combine the strengths of a mathematical model normalization together with a transfer function measurement normalization technique. Therefore the system uses an unsolved *Z* domain mathematical model as a target measurement system. The measurement is performed by inputting an impulse to the mathematical model and obtaining its maximum through the realization of a measurement on its output.

It is known from Fourier theory and linear system theory that:

$$i(t) = FFT^{-1}(H(w)) \qquad (3.)$$

where $i(t)$ is the output impulse response of the system, $FFT^{-1}(t)$ is the inverse Fourier transform and $H(w)$ is the transfer function of the system under study. By applying the following identity, where $f(t)$ represents an arbitrary time domain function,

$$f(t) = FFT^{-1}(FFT(f(t))) \qquad (4.)$$

and given that the input $x(t)=\delta(t)$ where $\delta(t)$ is an impulse then we can say that $y(t)=i(t)$, therefore:

$$H(w) = FFT(y(t)) \qquad (5.)$$

Thus the transfer function of a complex system whose input is an impulse response is given by performing the FFT of the output.

In other words the normalization constant can be found by applying an impulse to a mathematical model of a system, such as a *Z* domain function. Then a simple FFT is applied to the output. The resulting output can now be searched for the maximum value. In practice, only searching half the FFT data is necessary. The inverse of the obtained value is the normalization constant to be applied to the input.

The algorithm for implementing the automatic maximum gain normalization technique is presented in Figure 5. In a standard system, the user interface would be connected directly to the audio processing device. For demonstrating the algorithm, we have detached the user interface and stored the corresponding coefficients coming from the interface in a memory block called the fade in parameters block. This memory block sends the coefficients to the audio processing device once the normalization constant has been found. The coefficients together with the normalization constant are transferred using a linear interpolation algorithm that ensures a soft, modulation-free transition to the next system state. An advantage of the user interface detachment is that the method can be implemented on analogue systems by interfacing the analogue user interface with analogue to digital converters and by transferring the results to the audio device using digital to analogue converters.

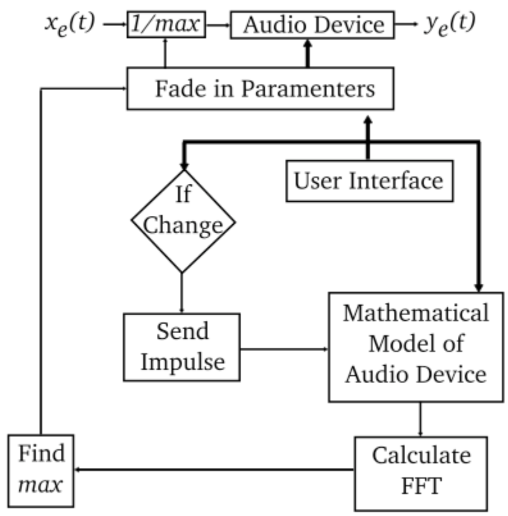

Figure 5 Algorithm of the proposed normalization technique using a truncated impulse response.

The algorithm sends an impulse to the mathematical model every time a change in the user interface has been detected. This ensures a correct normalization every time the linear system state has changed. Thus it is possible to calculate correctly the normalization value even if the transfer function order changes, for example when bypassing certain sections of an equalizer or even if the system design has changed, such as changing a filter in real time from a peak/notch to a shelf filter.

One of the advantages of this method is that it can be implemented either at clock speed or at sample rate speed. It also offers a more general solution to linear system normalization. The only section of the algorithm that needs to be revised if the linear system is changed is the memory sector containing the mathematical model. This gives the automatic maximum gain normalization technique the capabiliy of being implemented as a solid-state chip, which can be interconnected to memory containing the model.

3.1. Implementation

This technique has been implemented on a full parametric equalizer, Figure 6. The implementation uses six biquadratic filters. One of them is a low pass filter, another is a high pass filter and four of them are full parametric filters. The low and high pass filters have user frequency selectivity and the last four have frequency gain and quality factor (Q) user parameters. Also, the two outer parametric filters can be swapped between a peak/notch filter or a shelving filter. Every time a filter is modified, the coefficients driving the transfer function of the system change. Therefore a new normalization value is derived for every parameter change. The equalizer has the possibility of individually bypassing the high pass filter, the low pass filter, and the parametric filters. The compensated gain in dBfs is displayed at all times. A bypass button prevents the automatic maximum normalization technique for comparison purposes.

The mathematical model is given by equation 6. It is simply the unsolved Z domain transfer function of six biquadratic filtes in series, one per filter in the implemented equalizer, where the coefficients can be positive or negative. The FFT frame size used to implement the algorithm was 1024 and no windowing was used in order to minimize amplitude errors.

$$H(z)=\frac{(a_1+b_1z^{-1}+c_1z^{-2})}{(1+d_1z^{-1}+e_1z^{-2})}\cdot\frac{(a_2+b_2z^{-1}+c_2z^{-2})}{(1+d_2z^{-1}+e_2z^{-2})}\cdot\frac{(a_3+b_3z^{-1}+c_3z^{-2})}{(1+d_3z^{-1}+e_3z^{-2})}\cdot\frac{(a_4+b_4z^{-1}+c_4z^{-2})}{(1+d_4z^{-1}+e_4z^{-2})}\cdot\frac{(a_5+b_5z^{-1}+c_5z^{-2})}{(1+d_5z^{-1}+e_5z^{-2})}\cdot\frac{(a_6+b_6z^{-1}+c_6z^{-2})}{(1+d_6z^{-1}+e_6z^{-2})} \quad (6.)$$

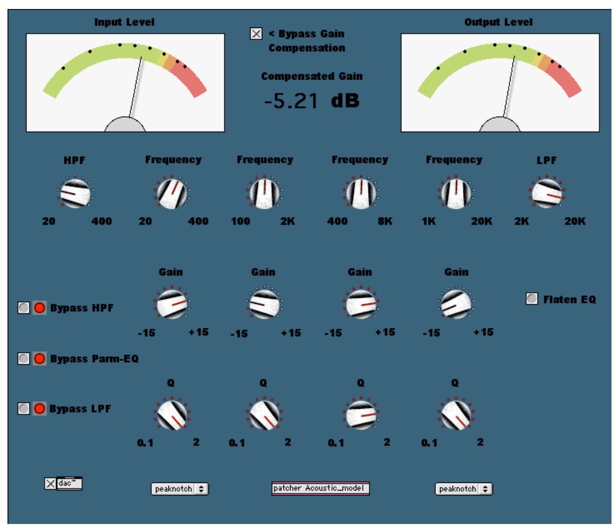

Figure 6 User interface of the implementation of the proposed normalization technique on a six biquadratic filter.

4. RESULTS

Open loop source independent measurements were performed for the implementation of the method on a six biquadratic parametric filter implementation. Measurements of the resulting transfer function were made using a sample rate of 44100 with a fixed point per octave FFT with a frequency resolution of 24 points per octave with a Hanning window with 32 vector averages.

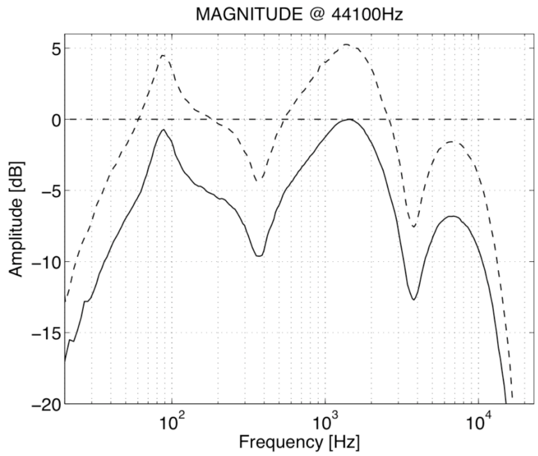

Figure 7 Transfer function of a un-normalized and a normalized response. The dash-dotted (-·-·-) line represents the threshold for maximum gain before feedback, the dashed line (- - -) represents the transfer function of a non-normalized acoustic system and the full line (——) is the transfer function after applying the normalization method.

Several boost and cuts corresponding to the equalizer user settings presented in Figure 6 have been plotted on Figure 7. The dashed line represents the non-normalized response of the equalizer while the solid line represents the normalized transfer function. The solid line has been successfully normalized below the 0dB threshold line. This means that boost functionality on the equalizer is still available relative to the normalization value and does not contribute by adding gain to the overall transfer function of the system. The overall compensation applied to the equalizer for these settings was -5.21dB.

It was also found that for low frequencies the lower frequency resolution below 400Hz could be affected if the Q of the filter is high. This is because the frame size truncates the impulse response of the system under study, causing loss of low frequency information. The error plot of gain normalization vs. Q is presented in Figure 8. It can be seen that the higher the Q, the higher the error. It can also be seen that the error changes in an exponential manner with respect to Q. This means that the error in estimation of the maximum of the transfer function is only significant for very strong filtering of very low frequency content.

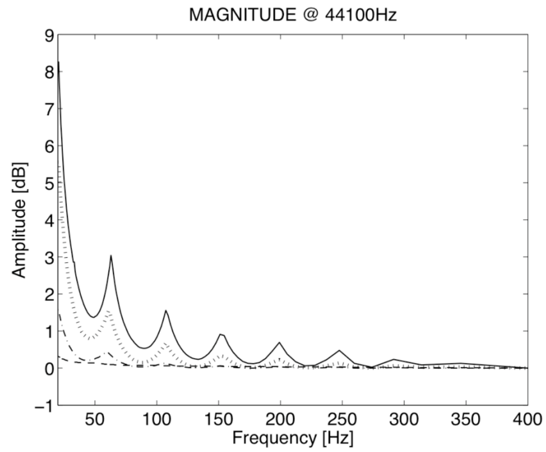

Figure 8 Error due to filter Q for a frequency range of 20Hz to 400Hz. The full line (——) is error for Q=2 (knob at full right position), dotted line (···) is error for Q=0.995595, dash-dotted (-·-·-) line is error for Q=0.371429 (knob at center position) and dashed line (- - -) is error for Q=0.1 (knob at full left position)

This particular low frequency error can be counteracted by using an inverted multiplying mask which matches the error plots presented Figure 8. On the other hand, using a constant Q transform might offer a more generalized solution. This remains a subject of future research.

Software simulation based on a single feedback path model like the one shown in Figure 1 was implemented. The model takes into account temperature to calculate the speed of sound and uses the inverse square law to determine the delay and amplitude of the feedback path contribution to the system. Under this condition the system behaved as expected, avoiding howling, for frequencies above 400Hz. After diminishing the overall electronic transfer function gain by 6dB the system performed as expected for all frequencies. This was attributed to the error associated with the use of high Qs in the low frequency range.

Laboratory tests on a real acoustic system were also performed. The experimental set-up and recording environment are shown in Figure 9. A self-powered studio monitor playing wideband-recorded music was used as a source. The speaker was placed 10cm away from an omni-directional flat frequency response microphone. Care was taken to keep the source level set such that microphone diaphragm distortions are avoided. The microphone was then connected to a soundcard interfaced to the software containing the automatic normalization parametric equalizer implementation. The output of the system was connected to a line driver to control the overall amplification gain of the system. Finally a self-power studio monitor was placed at 160cm from the microphone capsule. This speaker was used as the main sound reinforcement speaker. Care was also taken to avoid electronic and acoustic distortion over system.

Figure 10 Acoustic measurement setup.

While the equalizer remained flat, the system was driven to the marginal state of maximum gain before feedback. Afterwards, numerous boost and cuts were applied to the equalizer. Compensations of up to -50dBs were achieved without howling. It was observed that only a 3dB margin was required for avoiding howlback due to artifacts introduced by high Qs on the low frequency range. This is better than expected by simulation. It is thought that this is due to the room acoustics, which caused a 3dB destructive contribution to the feedback effect compared to an ideal constructive 6dB contribution achieved during the single path simulation using software.

5. CONCLUSIONS AND FURTHER STUDY

A normalization technique which prevents feedback has been introduced. The method performs real-time normalization of the gain of a changing linear system to stop it from going beyond the maximum gain before feedback threshold. Simulations and acoustic tests implemented on a six biquadratic parametric filter implementation have shown its suitability for use in sound reinforcement applications.

Further improvements to reduce the error in low frequencies due to impulse truncation must be performed. A constant Q implementation of the algorithm might solve this problem. Implementation of a similar system that normalizes phase in order to prevent feedback between several sources could also be implemented using a similar approach.

REFERENCES

[1] S. H. Antman, "Extension to the theory of howlback in reverberant rooms.," *Acoustical Society of America,* vol. 2.14, 2.7; 5.13, p. 399, 1965.

[2] P. d'acht', "Understanding Acoustic Feedback; Proyect Larsen Forever, 2008, www.artcontemporain.lu/larsen/larsen.htm."

[3] D. Davis, and E. Patronis Jr., *Sound System Engineering*, Third Edition ed., 2006.

[4] R. C. Cable, "The practical applications of Time-Delay Spectrometry in the Field," *Journal Audio Engineering Society,* vol. 28, pp. 302-209, May 1980.

[5] J. Meyer, "Equalization Using Voice and Music as the Sources," in *76th Audio Engineering Society Convention* New York, 1984.

[6] B. McCarthy, *Sound System Design and Optimisation, Modern technologies and tools for sound sytem design and aligment*: Focal Press, 2007.

[7] S. Kamerling, and etal., "A New Way of Acoustic Feedback Suppression," in *104th Audio Engineering Society Convention*, 1998.

[8] D. Troxel, "Understanding Acoustic Feedback & Suppressors," Rane Corporation 2005.

[9] J. Meyer, "Precision Transfer Function Measurments Using Program Material as the Excitation Signal," in *11th Audio Engineering Society Iternational Conference*.