Institute of Acoustics

# HOWL-ROUND DETECTION AND CONTROL USING ADAPTIVE FILTERING TECHNIQUES

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## INTRODUCTION

Howl Round is a perennial problem in sound reinforcement systems and is often aggravated by the need to provide automatic control of the system. One solution is to provide gain control circuits which always ensure that the system never goes into oscillation even under worst case conditions. Unfortunately this may result in a sound reinforcement system which does not provide a high enough level of sound in some circumstances.

A skilled P.A. operator however will equalise the system to both reduce feedback and provide a high quality sound to the listener. If the system begins to show signs of oscillation the operator would then reduce the gain of the system (possibly on a per microphone basis) to remove the problem.

There are existing commercial systems which attempt the model the behaviour of a skilled P.A. operator by detecting the presence of feedback and then reducing the system gain to remove it. The problem with these systems is that of detecting the presence of feedback in the audio signal. One system the author knows uses several phase lock loops as narrow band tracking filters to extract sinusoidal signal which might represent a feedback component. It then uses some complex decision process to decide whether feedback is present and thus whether to reduce the gain of the system. Needless to say the complexity of this approach is reflected in the cost of the device.

Another approach is to attempt, via adaptive digital filtering, to model the feedback path from speaker to microphone so that the unwanted feedback signal can be cancelled out by the output of the adaptive filter. Unfortunately this requires a very large adaptive filter to model the reverberant decay of the room.

Frequency shifting is another solution which has been proposed but this tends to cause problems with the quality of the audio signal, especially when music is being reproduced.

This paper describes a technique for automatically detecting and removing howl round by using a simpler form of adaptive filter which can be realized fairly cheaply. The rest of the paper first describes the theoretical aspects of howl round and the adaptive filter. The paper then describes an implementation of the idea and discusses the results and problems which occurred. The paper concludes with suggestions for both further improvement of the device and other possible applications.

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## THEORETICAL BACKGROUND

Figure 1 shows the two main paths of sound feedback to a microphone from a loudspeaker. These two paths are:

a) The direct path from loudspeaker to the microphone.
b) The reverberant field due to the loudspeaker output.

One can control path (a) to a large degree by suitable placement of appropriate type of microphone. However, path (b) is determined by the reverberant field in the room. Although this can be reduced via the use of directional loudspeakers there is a limit to what is practically; or economically achievable. The reverberant field represents the bottom line for feedback control using techniques which do not attempt to cancel its contribution. The reverberant field is also the contribution which must be cancelled if a signal cancellation technique via the use of adaptive filters is envisaged and this is why such filters need to be long (a significant part of $T_{60}$) to be effective.

The contributions of paths (a) and (b) to the system can be modelled as shown in Figure 2. Figure 2 models the system as an amplifier (gain A) with frequency dependent feedback $H_{(jw)}$. From this model we know that the system will oscillate if two conditions are met

1) $|AH_{(jw)}| = 1$

2) $\Phi(AH_{(jw)}) = 2h\pi \quad h = 0, 1 \ldots\ldots \infty$

Conditions 1 and 2 are known as the Gain and Phase conditions respectively.

The phase condition is dependent on the distance the sound has to travel before entering the microphone. As this is often several wavelengths it results in the system being able to oscillate at several possible frequencies. In theory if one could arrange the loop gain of the system to be <1 at these frequencies one could avoid feedback. However in a real environment there are many possible reflection paths so that the density of possible oscillation frequencies is too high to allow this approach. Frequency shifting can be viewed as a technique for controlling feedback by modifying the phase condition as a frequency shift imposes a time varying phase shift on the system. The effect of this is to dynamically change the phase condition so that the possible frequencies of oscillation vary at a rate which prevents the oscillation from building up. However in a situation with a high density of possible resonances the efficacy of the system is reduced.

The gain condition however is dependent both on the absorption characteristics of the room and on the frequency response of the P.A. system. If one considers a single acoustic propagation path then its frequency response will be a gradually varying function of frequency which can be readily equalised to a flat condition. However, in a real acoustic environment there are many acoustic paths and these will reinforce or cancel each other as function of both frequency and microphone location. One can equalise the gross variations in frequency response and so improve the feedback margin. But there is still the possibility of multipath causing a set of resonances which compromise the margins.

It would seem therefore the best one could do would be to have a system which could adaptively reduce the gain of the system when feedback was detected. Ideally this would be in the frequency region of the feedback to begin with and then over a broader frequency range if required. It would also be good if the system was cheap enough to be used on a per microphone basis.

## ADAPTIVE LINE ENHANCING

Figure 3 shows the block diagram of an archetypal noise cancelling filter. The reference is filtered by an adaptive filter so that when it is subtracted from the input signal the output of the system is minimised. Thus if the reference input is a sine wave at the same frequency as an interfering tone in the input (e.g. mains hum) then the adaptive filter will adjust itself so that the phase and amplitude of the reference source is identical to the interference source. This will result in the output of the adaptive filter being a signal identical to the interfering tone and thus the output of the subtractor will be the desired signal minus the interfering tone. We can model feedback as an interfering tone but, unfortunately, we cannot provide a reference input. The solution is to model the desired signal as random noise, in an information theoretic as opposed to an aesthetic sense. If this model is accurate then a delayed version of the input signal will not be correlated with the undelayed version. Feedback however will exhibit long term correlation. The result is that the necessary reference input can be derived from the input signal by delaying it as shown in Figure 4. For music a delay of about 50ms seems to be adequate to decorrelate the signal. This structure is known as the Adaptive Line Enhancer (ALE) because the output of the filter is an estimate of the tone present in the signal. The output of the substractor in this structure is the signal without the tone and thus represents the signal free from feedback.

## IMPLEMENTATION

The above system was implemented as shown in Figure 5. The delay of 50ms was provided by a bucket brigade device, the multipliers were implemented using multiplying DACs, and the adaption was done by using the exclusive or of the zero crossings of the signals to control the up/down counters which set the weights for the DACs.

The system provides a tunable notch by adjusting the phase and amplitude via two weights, on the delayed signal and a 90$^{o}$ phase shifted version of the signal. Strictly speaking this provides a tunable comb filter but it was simple to implement. The system successfully detected and removed feedback. However because of its comb filter response it was less effective than it could have been. However it did demonstrate the feasability of the basic idea.

## FURTHER WORK

The technique of using a delay to decorrelate the signal is useful for both detecting feedback and providing an estimate of the reverberant field level. A simpler system could use the delay and phase shifts to provide a detector of feedback as shown in Figure 6. The output of this detector will normally be at the level of the reverberant field but the presence of feedback will raise that level and so can be detected and used to reduce the gain of the system. The fact that the detector shown in Figure 6 gives an estimate of the reverberant field of the environment also suggests its use as an adaptive threshold control for gated microphone systems or noise gates. The adaptive filter could be improved as shown in figure 7. This structure actually implements an adaptive notch filter, as opposed to an adaptive comb filter, it also removes the delay line from the signal path and so improves the potential quality of the system. The adaption algorithm however is more complex.

## CONCLUSION

It is possible to use an analogue implementation of the Adaptive Line Enhancer to provide an effective and economic means of detecting and controlling feedback in sound reinforcement systems.

## ACKNOWLEDGEMENT

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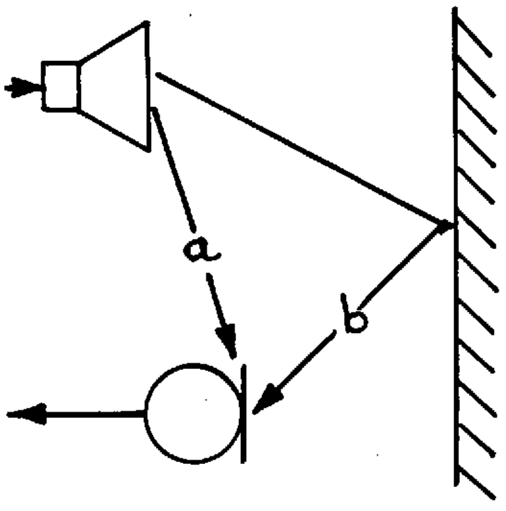


Figure 1

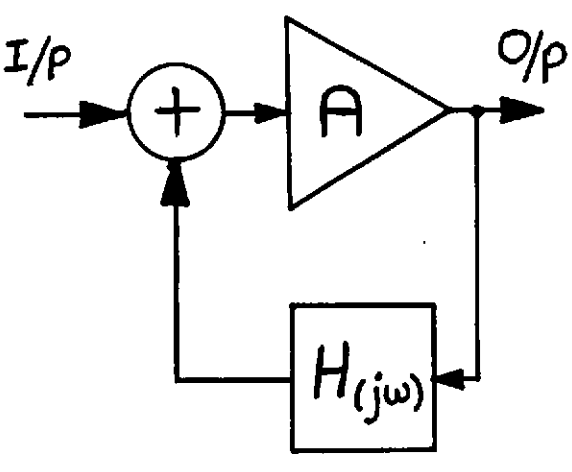


Figure 2

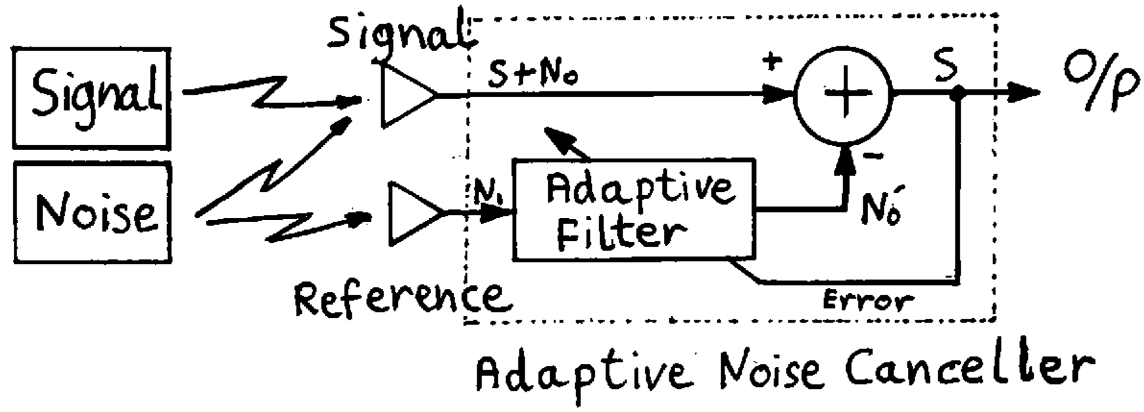


Figure 3

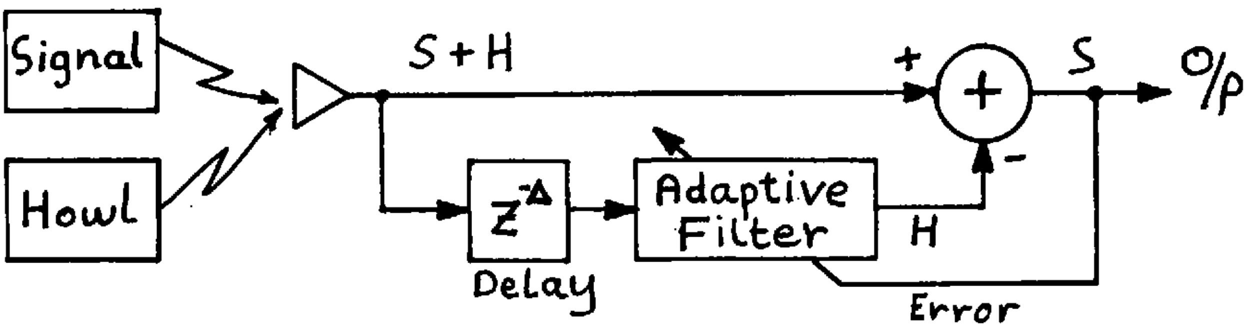


Figure 4

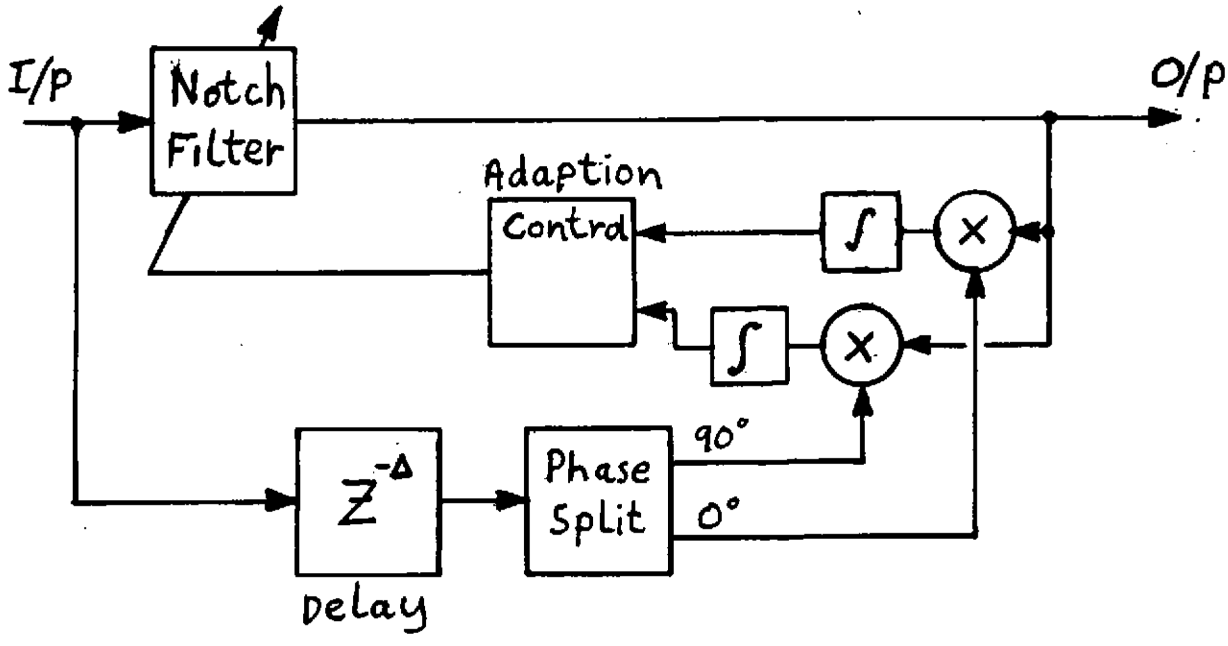


Figure 7

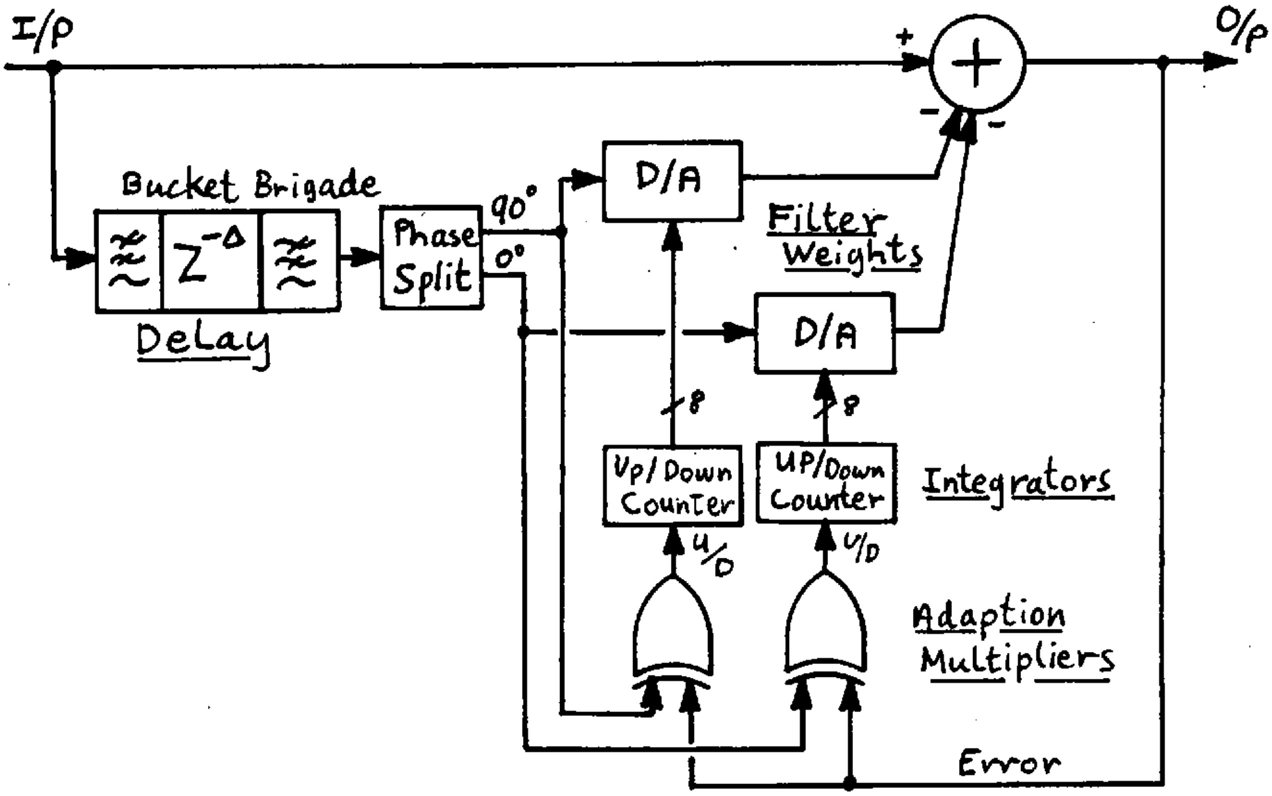


Figure 5

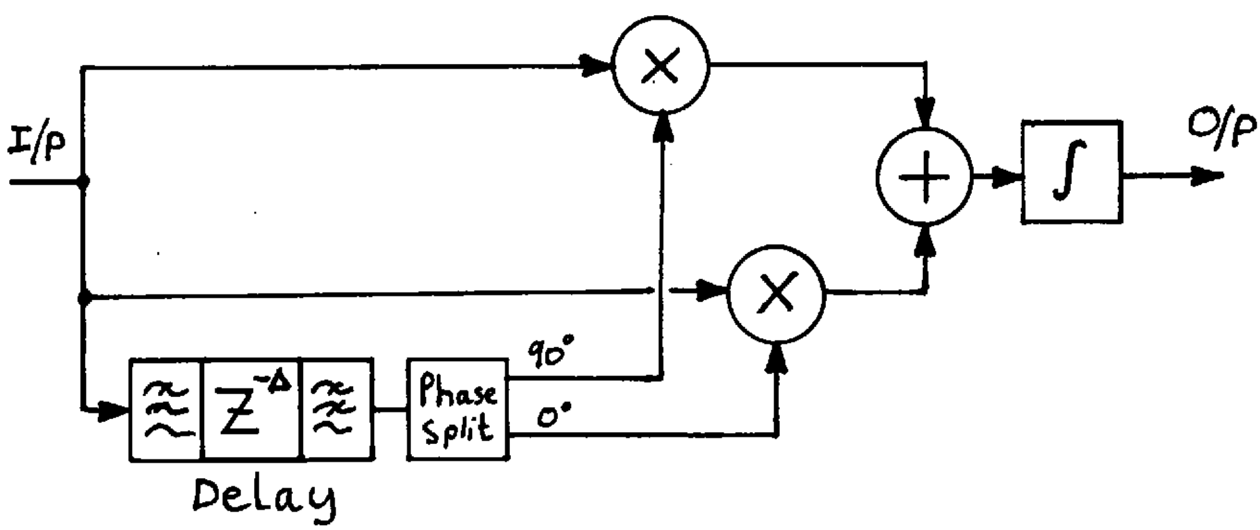


Figure 6