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## Active noise reducing headset – an overview

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### Abstract

Analogue active noise-reducing headset has been one of the most successful applications of active control of sound, with recent digital noise reducing headset promising even further improvement in performance and flexibility. This paper presents an overview of the active noise-reducing headset, from passive attenuation, through analogue feedback control to digital control. The effect of the delay introduced by the digital system on the performance with feedback and feedforward configurations is also discussed.

### 1. Introduction

Active headset is a noise-reducing headset that uses active noise control for the noise reduction. Active noise reduction in headsets dates back to the 1950's, where Simshauser, Hawley and Meeker [1]-[3] studied the use of analogue amplifiers to control a loudspeaker and a microphone placed inside the headset shell. Active headsets are used mainly in highly noisy environments to protect the user from the excessive noise. Such headsets usually use both passive and active attenuation. A good active headset will effectively combine low frequency active attenuation with high frequency passive attenuation to provide high attenuation of the external noise at a wide frequency range. Heavy noise reducing headsets that use passive and active attenuation to give maximum protection from high noise levels are used in aviation, military and industry. Lighter headsets worn by passengers in aircraft for example are becoming more popular recently. These are usually combined with audio headsets to provide a more comfortable ride while listening to the audio or video flight entertainment. This paper presents a technical overview of the active headset, from passive attenuation, through analogue feedback control, to digital feedback and feedforward control. Specific emphasis is placed on the delay introduced by the digital control system, and its effect on performance in the feedback and the feedforward configurations.

### 2. Passive attenuation

Passive attenuation is achieved when the headset shell is sealed to the head using an appropriate cushion therefore blocking the sound. Since the cushion needs to be soft or flexible to allow for a good fit and a tight seal, it also allows the shell to vibrate when exposed to external sound. The vibrations of the shell then radiate sound into the shell cavity, which is then perceived by the listener. This is illustrated in Figure 1. Heavier shells with stiff

cushions are more difficult to vibrate and therefore provide better passive attenuation. Shaw and Thiessen showed in 1962 [4] that the passive attenuation of the headset behaves as a second order mechanical system:

$$ATT = \frac{K_v}{K_v + K_c + j\omega R - \omega^2 M} \quad (1)$$

with  $M$  is the shell mass,  $R$  is the cushion damping and  $K_v$  and  $K_c$  the stiffness of the air in the shell volume and the cushion respectively. The passive attenuation of a typical headset is illustrated in Figure 2, showing about 15 dB of attenuation below the mechanical cut-off frequency, and an increasing attenuation above the cut-off frequency. Since the mechanical impedance due to the shell mass increases with frequency, the shell provides better attenuation at the higher frequencies. At low frequencies the attenuation is controlled by the stiffness of the cushion and shell cavity. In practice imperfect seal will degrade the attenuation at low frequencies, while high frequency dynamics of the shell and its cavity will affect the attenuation at higher frequencies. From Figure 2 it is clear that additional active attenuation at the low frequencies can improve the overall attenuation performance.

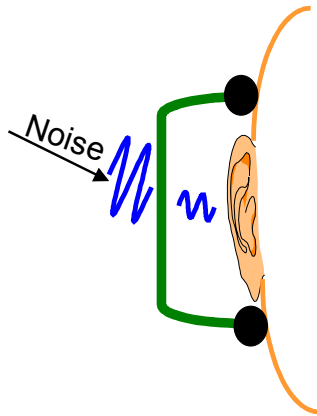


Figure 1. Passive noise reducing headset

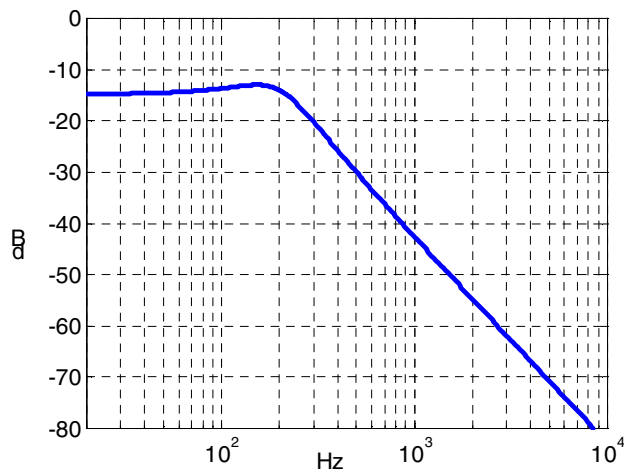


Figure 2. Simulated typical passive attenuation

### 3. Analogue feedback control

Meeker, Hawley and Simshauser [1]-[3] in the 1950's studied the potential of actively reducing the noise in a headset. A loudspeaker was placed inside the headset shell, and was driven by an amplifier that fed-back the signal from a microphone closely located to the loudspeaker, as shown in Fig. 3. At low frequencies, where the phase lag of the control system and transducers was relatively small, the loudspeaker produced sound that is similar but opposite in phase to the external noise, and noise cancellation was achieved. Nevertheless, the feedback control system, like any other feedback system, had to maintain stability, which was ensured by applying low control gains at the high frequencies. This brought the traditional design trade-off: performance vs. stability.

The headset feedback control system is presented by the block diagram in Figure 4, with  $P$  the plant, or system under control, which is the response from the loudspeaker input to the microphone output;  $C$  the analogue controller;  $d$  the external noise as measured by the microphone, also referred to as the disturbance signal; and  $e$  the signal at the microphone after control also referred to as the error signal. The error signal can be written in the frequency domain as the disturbance signal multiplied by the response of the closed-loop system:

$$e = d \frac{1}{1 + CP} \quad (2)$$

The response of the closed-loop system, denoted by  $S$  is referred to as the sensitivity function and is written as:

$$S = \frac{1}{1 + CP} \quad (3)$$

A simplified model of the plant response  $P$  can include the dynamics of the loudspeaker when radiating into a cavity which might not be perfectly sealed. In this case the plant response can be written as [5]:

$$P = \frac{\Phi}{Z_e Z_m + \Phi^2} \frac{aR_v}{1 + j\omega R_v C_v} \quad (4)$$

where  $\Phi$  is the force factor of the loudspeaker,  $Z_e$  and  $Z_m$  are its electrical impedance and mechanical impedance respectively,  $C_v$  is the acoustic compliance of the earshell cavity volume,  $R_v$  is the resistance of a leak and  $a$  is the area of the membrane. A typical magnitude response of  $P$  simulated using (4) is presented in Figure 5. A typical magnitude response of  $S$ , simulated using (3), the simulated plant response and a constant-gain feedback controller is illustrated in Figure 6. At low frequencies, where the open-loop gain is high,  $S$  is small and so large attenuation is achieved. At high frequencies the gain of the open-loop is small,  $S$  is close to unity and the controller has negligible effect. At the transition band, around 1kHz, the controller amplifies the noise. This amplification is inevitable for most practical headset systems, as suggested by the Bode integral [6]:

$$\int_0^{\infty} \log|S(\omega)| d\omega = 0 \quad (5)$$

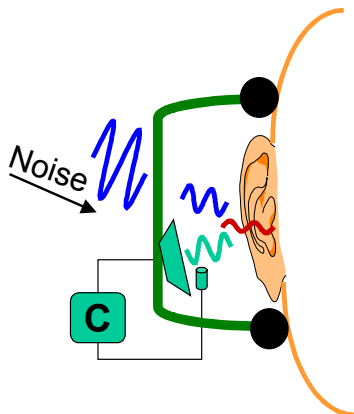


Figure 3. Active headset attenuation

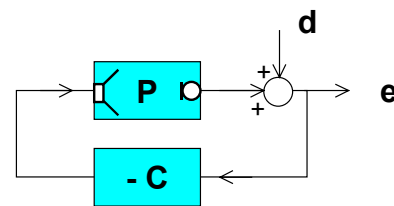


Figure 4. Feedback control diagram

This integral over the log magnitude of  $S$  over frequency give rise to the so-called “water-bed” effect, so that when pushing the attenuation curve down at one frequency range, the amplification will go up at another frequency range. Although amplification might occur at a wide frequency range, it is still reasonable as long as its level is small. It is interesting to note that although a simple model was used for the plant response and controller, the simulated attenuation is similar to that found in many commercial active headset systems.

Traditional controller design methods used modular filters elements, such as lead, lag and notch filters, appropriately tuned, to shape the response of the open-loop system such that a good performance is achieved with sufficient stability margins. In particular, stability margins are necessary at higher frequencies, where the response of the headset plant is uncertain due to manufacturing tolerances and variability between users. Recently, various design methods have been developed which attempt to provide a better trade-off between performance and stability by taking into account in more detail the plant uncertainty. Some of these methods such as H-infinity, Internal Model Control and Quantitative Feedback Theory have also been used in active headset applications [7]-[9].

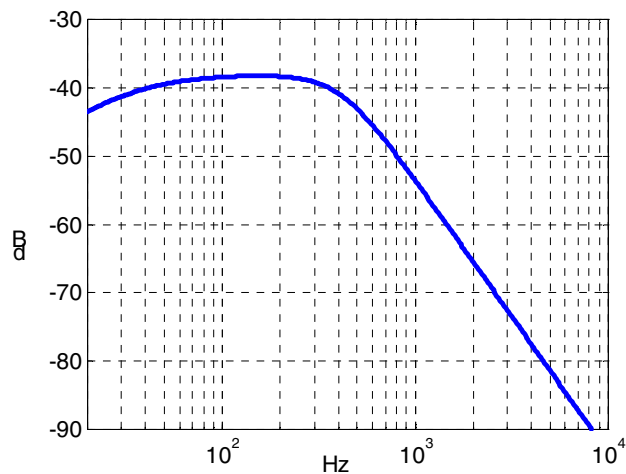


Figure 5. Typical magnitude response of a headset plant

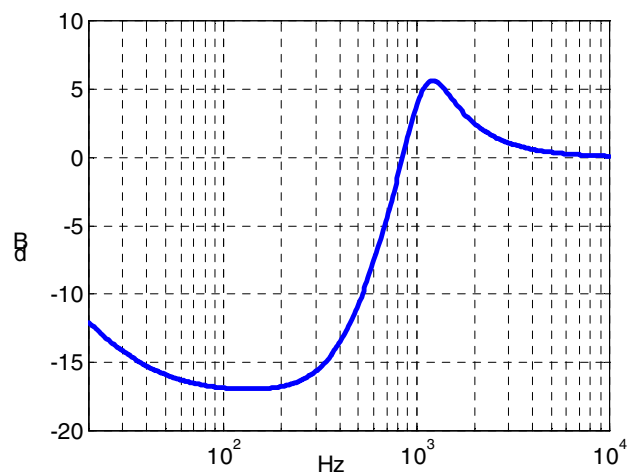


Figure 6. Typical active attenuation

## 5. Digital feedback control

The feedback controller presented above can be implemented using a digital system, with the advantages of implementation accuracy, flexibility, and the potential use of adaptive filters. Digital controllers, however, will be subject to additional delay due to sampling delay in the digital processor and the digital to analogue converters, and the phase delay of the low-pass filters. This delay can affect performance by limiting the control bandwidth, for example. To illustrate the effect of delay on the performance of an active headset, a simulation which used the plant  $P$  as in (4), with an additional delay, and a constant-gain controller designed to prevent any amplification over 6dB, is presented in Figure 7. As can be seen both control bandwidth and attenuation level are degraded with increased delay. Therefore, to achieve best performance delay must be minimized. This can be achieved by using a very high sampling frequency but will usually require powerful DSP processors and increased cost. Over-sampling can be used to reduce some of the delay [10].

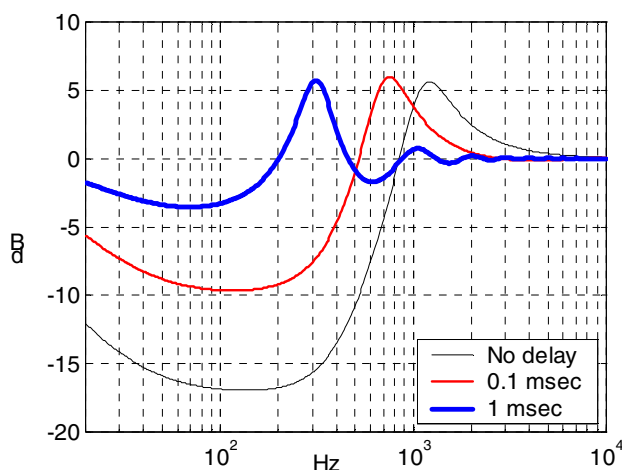


Figure 7. Attenuation with a feedback controller for a plant with additional delay of 0.1 msec and 1 msec.

## 5. Digital feedforward control

The application of adaptive feedforward systems in active noise control has been demonstrated in the 1980's by Roure, and Eriksson and Allie [11],[12]. A similar configuration can be used in an active headset, as shown in Figure 8, where an external microphone detects a reference of the noise, which is then filtered by the digital controller and transmitted to the loudspeaker. The internal error microphone is then used to tune or adapt the digital filter such that the overall level at this microphone is minimized. The LMS algorithm [13] is widely used for the adaptation of the digital filter. The schematic diagram in Figure 9 shows how the reference microphone detects the incoming noise, which is then filtered by the adaptive filter  $W$  before being transmitted to the loudspeaker. The error microphone detecting the total noise is then used to adapt the adaptive filter in order to minimize the mean square of the error. The following equation shows the adaptation rule as used in the Filtered-X LMS algorithm [13]:

$$w_i(n+1) = w_i(n) - 2\mu \cdot r(n-i) \cdot e(n) \quad (6)$$

where the coefficient  $i$  of the adaptive FIR filter  $w$  is updated every sample with the product of  $\mu$ ,  $r$  and  $e$ . Signal  $r$  is the filtered-X signal, calculated by filtering signal  $x$  with a model of the plant  $P$ .

Active feedforward control of sound rely on the timely detection of the reference signal, which needs to be filtered and transmitted to the loudspeaker in time to cancel the propagating primary noise. Excessive delay in the digital control path means that the cancellation signal will arrive too late to perform cancellation. If the total electric delay exceeds the acoustic delay from the reference microphone to the loudspeaker, then the optimal filter will be non-causal, and prediction will be required to attenuate the noise signals. In this case only band-limited or predictable signals can be successfully attenuated. Due to the relatively small dimensions of a headset system, the acoustic delay is usually relatively small, and would be expected to be smaller than a typical delay of a digital control system, therefore limiting the performance with broadband noise signals. Furthermore, the primary noise may not always propagate from the reference microphone to the error microphone. Nevertheless, broadband noise can still be reduced using a feedforward system, since the sound it transmitted in practice through the passive headset mechanism resulting in additional acoustic delay, as illustrated below.

The external and internal microphones in a typical feedforward headset system will be placed only few centimeters apart, which in open-air propagation is equivalent to an acoustic delay of 0.1-0.2 msec. However, in a closed-shell headset, the external sound propagates through the passive mechanism, and is subject to additional group delay. The phase response of the sound from outside the earshell to its inside, as in (1), is shown in Figure 10. Form the phase response, the group delay can be calculated as [9]:

$$\tau_g = -\frac{d\phi(\omega)}{d\omega} \quad (7)$$

and is presented in Figure 11. As can be seen the group delay is about 0.8-2 msec at frequencies below the mechanical resonance frequency, and reduces to zero above that frequency. This additional group delay could allow additional electronic delay without much reduction in performance. It is expected therefore that a digital feedforward controller will achieve some reduction of broadband noise at the low frequency range and not only narrow-band or tonal noise.

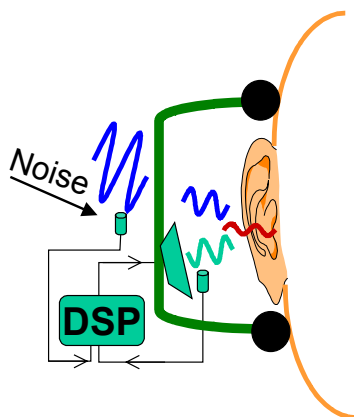


Figure 8. Digital feedforward headset system

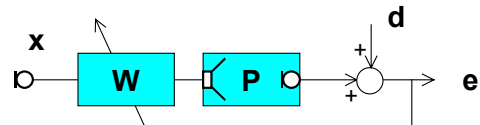


Figure 9. Feedforward control diagram

Experimental verification of this result has been performed by measuring the cross-correlation between the outer and inner headset microphones when the earshell was closed tight to the head and when it was open with large leaks allowing direct sound propagation. The primary noise was excited by a loudspeaker facing the side of the head. Figure 12 illustrates that the peak in the cross-correlation function occurs after about 0.25 msec when the earshell is open, and about 0.5 msec when the earshell is closed, showing the increased delay due to the mechanical response of the earshell.

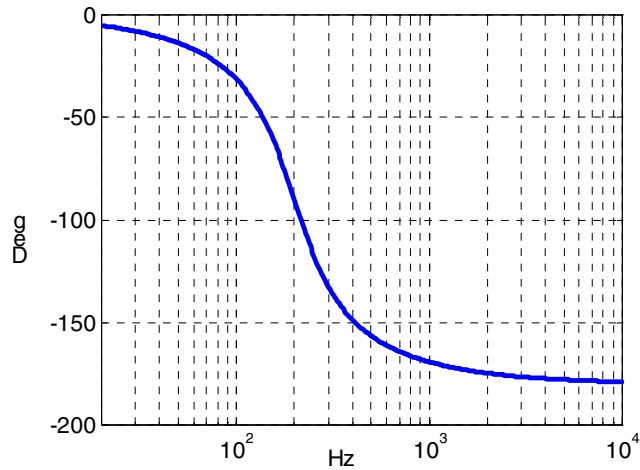


Figure 10. Phase response of the passive sound transmission through the headset

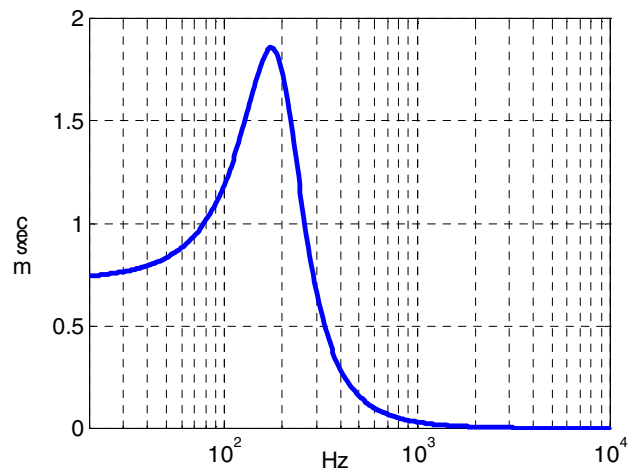


Figure 11. Group delay of the feedforward acoustic path

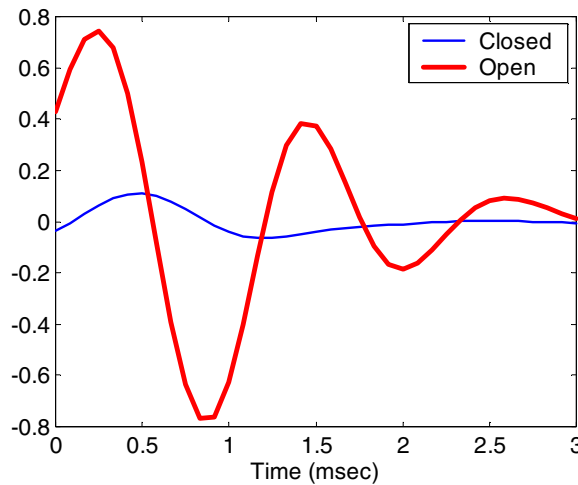


Figure 12. Measured cross-correlation between the external microphone and the internal microphone with the headset earshell closed tight to the head and open with air gaps.

## 6. Combined feedback and feedforward

The feedforward and feedback systems described above can be combined into a single system. The benefits from combining feedback and feedforward active control has been demonstrated by Tseng, Rafaely and Elliott in 1998 for enclosures [15]. Combining analogue feedback and digital feedforward has been demonstrated for headsets by Winberg *et al* and Carme in 1999 [16],[17]. In this configuration, the analogue feedback controller  $C$  is combined with the adaptive feedforward controller  $W$  by adding their control outputs at the loudspeaker input as shown in Figure 13. One way to view the new system is by considering the plant together with the analogue controller as the new plant, controlled by the digital system. The analogue controller therefore controls plant  $P$ , while the digital controller controls plant  $P_d$ , which also includes the analogue control loop.

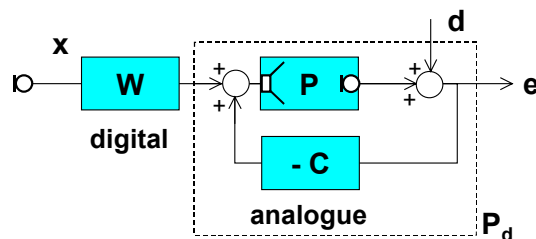


Figure 13. Combined system



## 7. Conclusions and future directions

Improvements in the response of transducers, better passive designs and better analogue controllers, can improve the traditional aspects of active headsets. However, with DSP increasing in power and decreasing in cost, digital control can produce better performance by attenuating broadband noise, and tracking fast changes. More powerful algorithms can be used which improve performance and tracking capabilities. These include frequency domain, sub-band and multirate systems which become increasingly popular in many applications [18],[19]. Also, the possibility of using a single microphone system or an adaptive feedback system as suggested in previous studies can produce a more compact system [20],[21]. With passive and combined systems operating, attenuation levels will improve. Nevertheless, noise perception through bone conduction imposes an upper limit on the headset attenuation, which is around 40 to 50dB.

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