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Digital Active Noise Reduction Ear Plugs

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FOR THE COMMANDER

MARIS M. VIKMANIS
Chief, Crew System Interface Division
Air Force Research Laboratory

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A digitally-controlled Active Noise Reduction (ANR) ear plug prototype system was developed. In contrast to available ANR headsets that implement fixed analog filters, the prototype defines a digital filter that is optimally defined for the user's current acoustical environment. An above-the-ear (ATE) and an in-the-ear (ITE) prototype ear plug system were developed and demonstrated. A passive headset is worn over the ANR ear plug to increase passive attenuation. A reverberant room was built and a KEMAR mannequin was used to test the ANR ear plugs in band-limited random noise with a spectrum similar to jet noise. Under these conditions, the ATE and ITE respectively gave 30 dB and 40 dB of overall passive attenuation and both gave an additional 10 dB of overall active attenuation. The ATE cancelled noise from 20 to 400 Hz and the ITE cancelled noise from 20 to 3 kHz. Currently only the ATE prototype is fully human compatible and capable of performing in 140 dB external sound levels. Previously-available ANR headsets fail when external sound pressure levels exceed 130 dB. Successful ANR is limited to low sound levels because the ANR system's controlled secondary sound source is normally required to operate in a linear regime while matching the amplitude of the primary noise.

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I. Introduction

A. ANR Concepts

Active Noise Reduction refers to the attenuation of "primary" sound by destructive interference with a controlled source of "secondary" sound. Secondary sound or "antisound" refers to sound originating from a speaker in an Active Noise Reduction system. Primary sound refers to sound originating from all other sources. Active Noise Reduction (ANR) is also referred to as *active* noise cancellation, attenuation, or control. Sound can also be reduced by the introduction of materials that reflect, absorb or impede the propagation of sound, and this is referred to as *passive* reduction, attenuation, or control. Passive Control issues related to this project are discussed in Section II. Distinct technologies have been developed for ANR headsets and earplugs according to whether the ANR control system is analog or digital, and according to whether the control system uses "feedforward" or "feedback" control [1, 2]. The work presented here describes the development of two types of ANR earplugs that incorporate an innovative digital feedback control system.

The frequency dependent electronic amplification between the input to the ANR control system and the driving voltage output to the secondary sound source is here referred to as the "ANR filter". In digital ANR systems, the ANR filter is composed of a digital filter in series with analog components. In analog ANR systems, the ANR filter incorporates only analog components. In existing *analog* ANR headsets the ANR filter may have a variable, self-adapting, overall gain control, but the ratio of the complex gains at two different frequencies remains *fixed.* Digital ANR systems are much more amenable for implementing techniques to adaptively adjust the frequency dependent gain of the ANR filter in order to optimize performance under varying conditions. *Feedforward*, adaptive, digital ANR headsets and earplugs use an "input" microphone to provide input to the controller and a second "error" microphone to determine the sound at a point where noise is to be cancelled. A single microphone performs both of these functions in adaptive digital ANR headsets and earplugs that are based on a controlled negative *feedback* loop.

The feedforward digital control system relies upon the primary sound inducing a strong *cross correlation* between the input microphone and the time delayed error microphone signals. Similarly, the single microphone feedback control system relies upon the primary sound inducing a strong, time delayed *autocorrelation* of the microphone signal. For both the feedback and feedforward cases, a good correlation after a time delay is important because the ANR system requires time to react to primary sound at the input microphone by delivering secondary sound to the error microphone. Let the primary sound correlation time scale be represented by τ_p, and the ANR system's reaction time scale be τ_s. It is desirable to have $\tau_p > \tau_s$ so that when the secondary sound arrives, the primary sound has not unpredictably changed (too much) from the time when the primary sound stimulated the secondary sound. As the quantity $\tau_p - \tau_s$ increases, the ANR filter may increasingly *add* to the delay time of the ANR system. This then increases the amount of variation with frequency that the complex gain of the ANR filter is capable of attaining. Any type of filter with frequency dependent discrimination, *e.g.* low-pass and hi-pass filters, causes a delay.

In the limit that the ANR filter is constrained to add no additional delay to the signal it processes, the ANR filter must have a constant gain as a function of frequency. Thus, the frequency dependent complex gain of an ANR filter that is desirable for good ANR performance will typically be approximated by a real (causal) ANR filter with increasing accuracy as $\tau_p - \tau_s$ increases. Note that periodic noise is relatively easy to cancel because $\tau_p = \infty$. Because it is often not the case that $\tau_p >> \tau_s$ in addition to it being desirable to choose fast microphones and speakers for ANR systems (to make τ_s small), it is also desirable to choose microphones and speakers with a flat amplitude response as a function of frequency. Sharp frequency dependent variations in these components will need to be compensated for by the ANR filter for consistent ANR performance across the frequency bands where these sharp variations occur. But the ANR filter's frequency dependent gain is unable to compensate well because of the handicapping constraint against adding more time to the response delay of the ANR system.

The preceding material introduced issues relevant to both feedback and feedforward digital control systems. At this point, attention is specifically focused on the ANR feedback control system. A fundamental limitation of a feedback controller is that it can only remove that portion of the noise that contributes to its own autocorrelation with a time delay at least as long as the response delays in the feedback control loop. Delays in the digital ANR feedback control loop include the electromechanical microphone delay, the computational time required by the digital signal processor (DSP) to implement the digital filter, the delays of the A/D and D/A converters, the delays of analog circuitry components such as the antialiasing and reconstruction low pass filters, the electromechanical speaker delay, and the acoustical transit time from the speaker to the microphone.

Random white noise (which is sometimes referred to as "true" random noise) is characterized by a "flat" acoustical power spectrum. Regardless of the center frequency, any two frequency bands having the same width contain the same amount of acoustical power. The overall sound level of random white noise cannot be attenuated at the microphone of a negative feedback loop because random white noise has no autocorrelation after a finite time delay ($\tau_p = 0$). After the time delay required for the feedback loop to react to primary sound it will be up to chance whether the updated primary and secondary sound add destructively or constructively.

Fortunately, in the real world noise is typically more bandlimited, unlike random white noise. Technically, acoustical airborne white noise is a theoretical construct that does not exist in nature, because sound can not have equal amounts of power in frequency bands of equal width with *arbitrarily* high center frequencies. Infinitesimally small bandwidths contain all the acoustical power of periodic noise. For non-periodic noise, many factors result in the spectrum being "unbalanced" or dominant in particular frequency bands. For example, it is typical for passive attenuation to increase at high frequencies, and for frequency bands favored by the acoustics of the noise source and external environment to contain most of the noise. These factors increase the tendency of noise to be dominant in particular frequency bands, and this coincides with a tendency to increase the time delayed autocorrelation of the noise.

In addition, for the work presented here, the ear sensitivity [3, 4] weighted noise is being minimized. A prototype ANR system was developed with a user friendly interface allows one to chose any one of several different ear sensitivity weighting currently used by audiologists. Typically the primary noise spectrum will be less flat after being weighted by the frequency dependent ear sensitivity. Considering the ear sensitivity weighting does not alter the autocorrelation of the primary sound. However, in this case there is still an enhancing effect resulting from the relevant spectrum being unbalanced. In principle, the overall *ear sensitivity weighted* SPL of random white noise *can* be reduced by a tightly coupled feedback loop that cancels sound in frequency bands where the ear is more sensitive and amplifies sound in other frequency bands.

A second fundamental limitation on the optimal cancellation of a feedback control loop is due to feedback instabilities. The controller is trying to process an input signal in order to force the input signal to zero. For a single microphone system to cancel noise perfectly, the controller must output the proper driving voltage to the speaker while the input signal is zero. Thus perfect cancellation would require an infinite gain. But because of delays in the feedback response, a feedback instability would be induced long before the gain of the controller reached infinity. Attempting to drive the microphone signal to zero results in an apparent paradox: typically the *best* ANR feedback system is very close to being the *worst* ANR system, because a high feedback gain results in being precariously close to going feedback unstable. When a person's ear has acoustical characteristics that deviate too far from normal, available analog ANR headsets can go unstable.

A third limitation of the negative feedback control loop is that there is some distance between the input microphone and the eardrum. These two locations will have different sound levels, due to the acoustical effects of the ear canal and ear drum. The ear canal and ear drum acoustical effects vary widely from person to person. The innovative feedback control system adaptively accommodates the manner in which a specific user's ear effects the ANR feedback loop, but does not yet take into account the acoustics between the ANR microphone and eardrum. While the current ANR system can be thought of as self-customizing for the current acoustical effects of the particular user's ear on the feedback loop, it is hoped that this customization can be extended so that the ANR system will also properly take into account the acoustical effects of the particular user's ear canal.

Efforts are proposed for future work that will enable the ANR system to measure, in a practical manner, parameters that are needed to model the user's current ear canal/drum acoustics and subsequently determine an appropriate ANR digital filter. This would enable one to attempt to directly attenuate sound at the eardrum, rather than attempting to attenuate sound at the ANR system's input microphone and hoping attenuation at the eardrum will follow. This work would involve making separate models for the primary and secondary sound in the ear canal. The discrepancies between the primary and secondary sound ANR microphone to ear drum transfer functions are not yet fully understood. In part the discrepancy could be due to the fact that primary and secondary sound take different paths to reach the ANR microphone and eardrum, and primary and secondary sound have distinctive near field effects on the ANR microphone. In addition to overcoming this third fundamental limitation (distance between ANR mic and ear drum), this future effort would also alleviate the second limitation (feedback instabilities). Because of the distinctive transfer functions of primary and secondary noise between the ANR microphone and eardrum, one would no longer be directly attempting to drive the input signal to zero, especially at higher frequencies.

B. Related ANR Work

Olson and May (1953) are credited for first applying a negative feedback loop to actively cancel unwanted sound [5,6]. Olson found it was best to place the microphone in close proximity to the speaker in order to minimize the delay of the feedback loop. While Olson was primarily interested in cancelling sound in a small vicinity around the microphone, researchers later applied the analog negative feedback loop to achieve global cancellation in the far field of low frequency sound traveling in one direction in a duct [7,8].

This duct research includes "conventional monopole" and "tightly coupled monopole" systems. For the conventional monopole system, the microphone is displaced away from the speaker in the direction that the sound is coming from. It is known [9,10] that overall attenuation of random white noise requires such a feedforward control approach to compensate for delays in the control system. More relevant to the research presented here is the tightly coupled monopole system, in which the microphone is placed directly over the secondary sound source (and not upstream). For example, Hong, Eghtesadi, and Leventhall [11] developed a tightly coupled monopole system and reported active attenuation of duct noise from 0 to 400 Hz that was as high as 30 dB at some frequencies. For this work only analog components were used to incorporate a desired frequency dependent gain in the feedback loop.

Andrea Electronics recently developed an adaptive, digital, feedforward system for an ANR earplug [12]. This approach was tested and the results are described in Section III. The use of an analog feedback control loop to cancel sound in an enclosed volume has been developed in many commercial applications. For example, the Bose Corporation ANR headset and the Sensor Electronics ANR earplug are described in various patents [13,14]. It is well known that for digitally controlled ANR systems where there is feedback, there are advantages to using a recursive IIR digital filter [15]. For the problem of cancelling sound in a enclosed volume, Billoud [16] used a single microphone and speaker in a digitally controlled feedback loop to simultaneously cancel multiple narrow bands of random noise from 0 to 500 Hz. The filtered-U LMS algorithm that Billoud used to define an adaptive IIR digital filter, and modifications there of, were investigated in this project. In experiments, we found this algorithm to be vulnerable to feedback instabilities. A modified version was tested that included online feedback loop calibration [17] and software to reject incremental updates to the digital filter that theoretically resulted in an insufficiently stable system. This modified version was stable, but the additional computational burden resulted in a low Nyquist frequency and poor ANR performance. Other researchers have patented alternative methods to stabilize the filtered-U LMS feedback control approach. However, this algorithm does not take the ear sensitivity into account. Also it is not clear how this algorithm could be extended to take the acoustics of the specific user's ear canal into account, so as to attempt to cancel noise at the eardrum instead of the ANR system's microphone.

In Section IV, an innovative algorithm is described for designing an IIR digital filter that will determine the frequency dependent gain of a digital filter in a negative feedback loop. The digital filter is designed in a manner to optimally reduce the ear sensitivity weighted noise at the microphone. Thus the amount of active attenuation will be greatest in frequency bands where sound is perceived as loudest to the human ear. The computation of the optimal digital filter takes into account the acoustics of the ANR system and the particular noise spectrum being encountered. The optimal digital filter is computed using a computation in frequency space, and then the resulting digital filter is implemented in real time. Another feature of the method is to incorporate a safety factor in this calculation, to insure that the feedback loop will not go unstable. Without this safety factor, instabilities might result when there are small inaccuracies in the calibrated response of the ANR system [18].

Other research in active noise control has based the design of a controlling digital filter on frequency space analysis, for a two microphone feedforward system. Several patents have been developed for digitally controlled systems that cancel periodic noise by taking a Fourier transform of a segment of input, applying a frequency dependent gain, and finally using an inverse Fourier transform to generate an output driving signal [19-21]. Applying such a system to cancelling random noise with a tightly coupled feedback controller is difficult since the finite time window required for resolution in Fourier space will cause a large delay in the response of the system.

A method presented by Ross did not involve calculating Fourier transforms in real time. The procedure of using a frequency domain calculation to determine an optimal digital filter for active noise control was introduced in the thesis, papers, and patent of Ross [22-25]. Ross used a computation in frequency space to design a two microphone feedforward controller that would cancel random sound traveling in one direction. By analyzing transfer functions, Ross calculated the gain needed in a system that uses the upstream microphone for input and produces a driving signal to the speaker on output. Subsequently, a calculation was made of the IIR digital filter coefficients which would realize this frequency dependent gain. This digital filter was then used to cancel random broadband noise from 0 to 500 Hz.

The work by Ross was specific to two microphone systems. For the two microphone feedforward system considered by Ross, the optimal frequency dependent gain can be computed algebraically. Applying a frequency space analysis to the tightly coupled feedback controller requires a very different calculation. As shown in this paper, the optimal transfer function for the tightly coupled feedback controller is not found directly. A function, the ear sensitivity weighted sound pressure level, is represented by an integral over frequency space. Here the integrand depends on each of the coefficients in the IIR digital filter. By numerically computing the minimum of this function of many variables subject to a relative stability constraint, the digital filter coefficients are determined directly, (*i.e.*, without determining the desired frequency dependent gain of the digital filter as an intermediate step).

Section V describes the prototype ANR systems developed for use in high noise environments. Band limited random noise with a spectrum similar to jet noise was of particular interest. Section VI describes the unique performance results that were demonstrated. A combined headset and ANR earplug was developed that was operational up to 140 dB. Another ANR earplug demonstrated simultaneous cancellation at all frequencies between

20 Hz to 3 kHz.

II. Equipment and Passive Control

The initial stages of the project involved the acquisition and assembly of needed experimental equipment. A reverberation room was built inside of a noise isolated room at the National Center for Physical Acoustics (NCPA). Noise having a spectrum similar to jet noise could be repeatably simulated within the reverberation room at levels up to 140 dB. A KEMAR mannequin was also acquired. The ANR systems developed at the NCPA reduce pressure fluctuations at the eardrum on the order of 50 dB, but do not reduce vibrations significantly. It was found that vibrations were distorting experimentally measured sound pressure levels obtained from 1/2" B&K microphones at the KEMAR eardrum. New ear drums were custom made to fit in the Zwislocki couplers of the KEMAR mannequin using Knowles EK-3132 microphones. The KEMAR was mounted on a vibration damped platform and the hollow interior of the KEMAR was filled with clay and sound absorbing materials to damp out vibrations and internal standing modes. The EK-3132 is rated to be 22 dB less sensitive to vibrations than the B&K microphones. Electronic components for the DSP system were assembled. This included a PC, Sonitech card, custom made integrated interface card (IIC), and many other miscellaneous items. The Sonitech card's capability to perform parallel processing on two TMS320C40 [26] DSPs was used to implement stereo operation of the ANR earplugs.

About two dozen different headsets and two dozen different earplugs were purchased. Rough comparisons of passive attenuation were made using the KEMAR mannequin in 140 dB jet noise. For the headsets, the David Clark 9AN/2 and Howard Leight Thunder 29 performed best. The 9AN/2 performed slightly better, but the Thunder 29 was easier to adjust. Ben Mozo conducted human threshold experiments and found that the David Clark 9AN/7 performed better than the 9AN/2, but the 9AN/7 model (which had very large earcups) was discontinued many years ago. The triple flange soft rubber ER4-14 Etymotic earplug and several slow recovery foam earplugs had superior passive control. The deciding advantage in favor of the ER4-14 is that it can be easily cleaned and is durable, which is of great practical importance for a relatively expensive ANR earplug.

Mozo tested earplugs alone and in conjunction with SPH-4 headsets [27]. Using earplugs alone, Mozo found the foam earplugs to be superior to the triple flange at all frequencies, with a maximum of about 10 dB superiority at 4000 Hz. However, when used in conjunction with a SPH-4 headset the differences were less noticeable and in five different military vehicles an estimated superiority in reducing the overall A-weighted sound was no more than 2 dB. Berger [28] tested an "LV" (large volume earmuff) David Clark 19A (similar to the 9AN/2) headset in conjunction with a "DI" (deeply inserted) foam earplug and another type of earplug (V51-R Harvard Psychoacoustics). Berger found the 19A/DI combination to have a maximum passive attenuation superiority of 15 dB at 250 and 500 Hz over the 19A/V51-R combination. However, the V51-R is inferior to triple flange earplugs.

Berger states that the passive attenuation of headsets and earplugs do not add linearly for two reasons: mechanical coupling and the bone conduction of the human skull. Berger's paper gives an estimate of the passive control limitations (of devices that do not cover the entire head) due to bone conduction. Bone conduction (BC) is not only a limitation on passive control, but BC can also be a limitation of ANR when a feedforward approach is used. Stimulus for vibrating the human skull may not be detected by the upstream input microphone of a feedforward ANR system. Because of the occlusion effect even a strong heartbeat, for example, can be relatively loud in an occluded ear canal. Sound beneath the earplug caused by BC is not amenable to active control using the feedforward approach, because the input microphone signal may not be well correlated with the vibrations of bones in the human skull. ANR performance can be sacrificed because BC degrades the correlation between the input and error microphones.

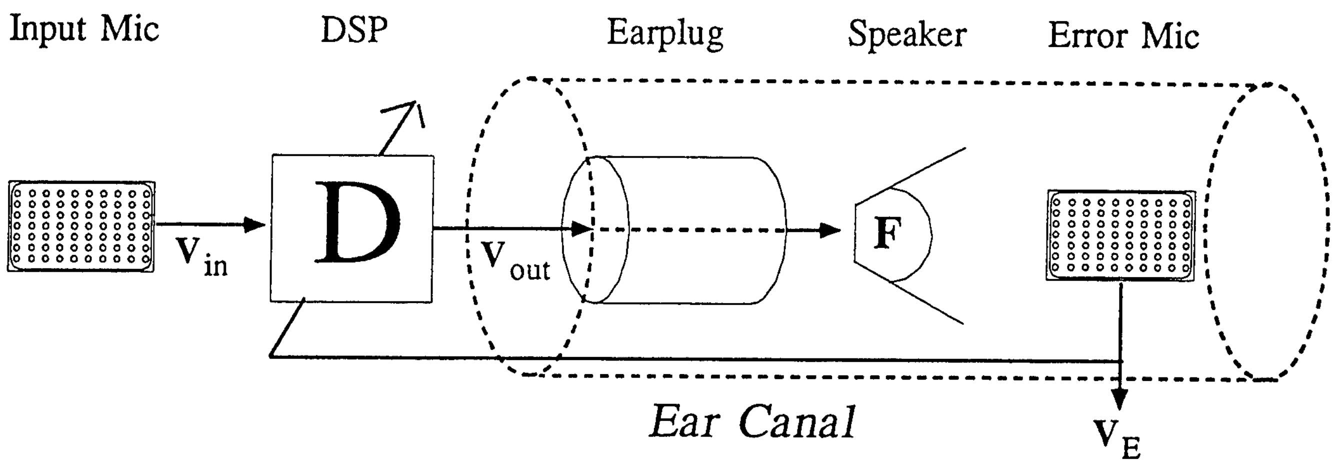
The ER4-14 fit especially well in the perfectly cylindrical KEMAR metal ear canal. The passive control of the two ANR earplug prototypes developed (ITE and ATE) was enhanced by placing some ear plug wax above the earplug. On the KEMAR, the ATE and ITE system's passive control exceeded Berger's "current" human BC limits at 2, 3.15, and 4 kHz by about 8 dB. Only the ITE prototype, which includes a 9AN/2 and ER4-14, exceeds BC passive limits at 6.3 and 8 kHz and the excess is about 10 dB. Therefore our KEMAR passive results are of questionable relevance for humans. This issue does not directly call into question the relevance of the result that the ATE prototype system on KEMAR can successfully operate in a linear regime up to 140 dB of jet noise, because typically most of the sound in external jet noise spectrum is in an octave band centered at at 1000 Hz, where passive control of the prototypes is below the human BC limits.

Our KEMAR tests showed the triple flange to have extraordinary attenuation from 0 to 100 Hz, and the lowest frequency that Mozo (and others) tested was 125 Hz. A decision was made to acquire additional 9AN/2 and Thunder 29 headsets in addition to ER4-14 and slow recovery foam earplugs. In addition to being incorporated into the ANR headsets/earplug systems, through out the course of the project these passive devices were essential for people carrying out experiments, since sound levels were as high as 100 dB outside the 140 dB reverberation room.

III. Testing of Feed-Forward Approach

The first generation of ANR ear plugs (ANREPs) relied on a digital feedforward controller using the conventional filtered-X LMS algorithm. A feedforward ANR earplug is schematically depicted in Figure 1. An "upstream" input microphone above the earplug generates input for the digital filter, and the coefficients that define the digital filter are slowly varied to minimize an "error" microphone below the earplug. There are several desirable conditions for this system to cancel random noise well. First, it is better if there is little feedback, *i.e.*, the secondary sound should be poorly detected at the upstream microphone. For the ANR earplug, this condition is well satisfied because the earplug's passive barrier reduces the amount of secondary sound reaching the input microphone. Second, the response of the two microphones should be strongly coherent when excited by primary noise. This condition was not perfectly met for the ANREPs tested in the KEMAR mannequin, but the condition was adequately satisfied for moderately successful ANR performance.

Figure 1. Schematic Depiction of ANR Earplug For Feed Forward Approach



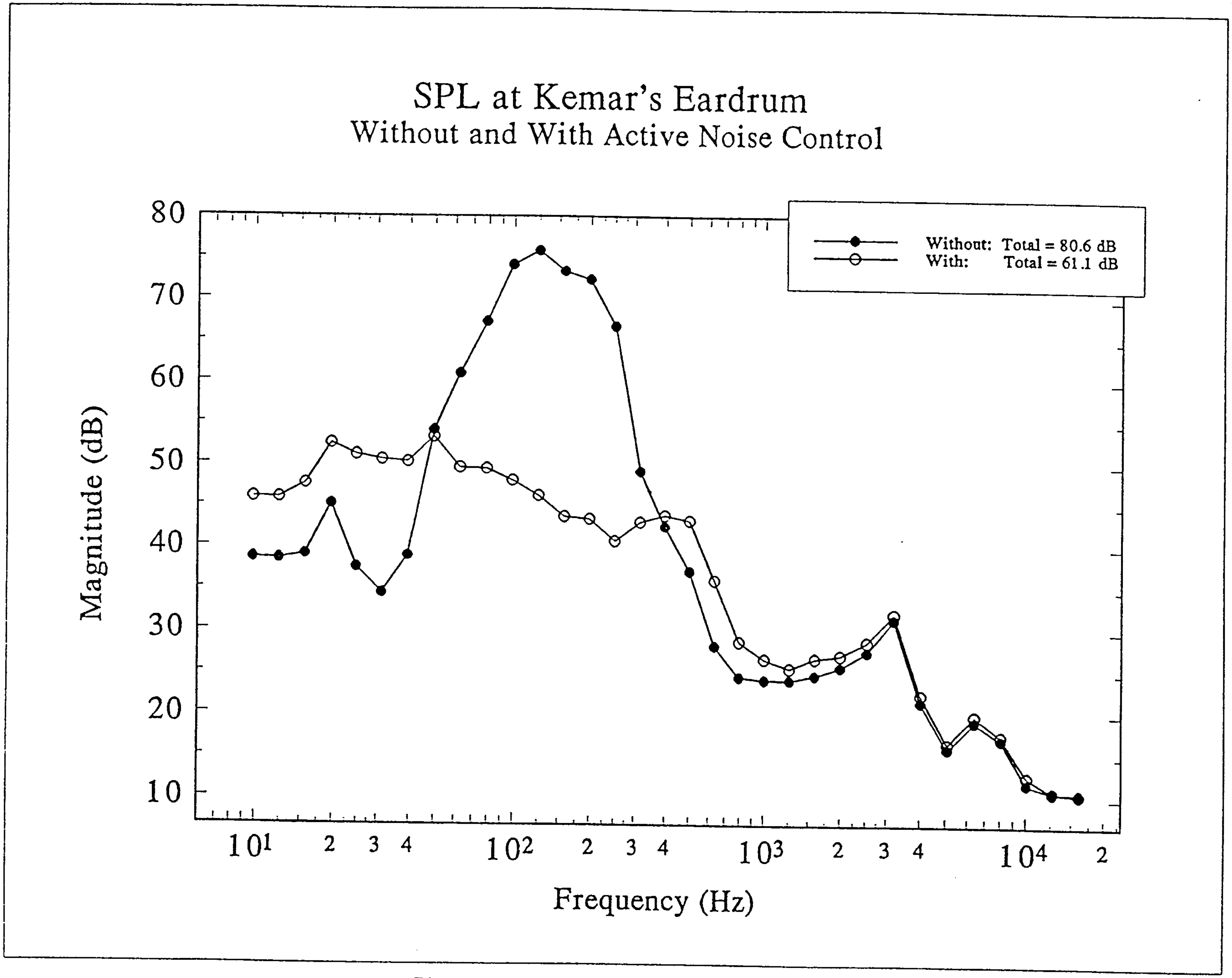
A third criterion has to do with timing issues that are associated with the constraint of causality. Let τ_p be the time required for primary sound to propagate from the upstream microphone to the error microphone. Let τ_s be the time required for sound passing over the input microphone to induce an electroacoustical response so that secondary sound reaches the error microphone. If $\tau_p >> \tau_s$, then the problems associated with the causality constraint can be alleviated. Unfortunately, for the feedforward ANREPs tested, $\tau_p \approx$ 100 usec $< \tau_s \approx$ 200 usec. Even with this unfavorable condition, the ANR system may work to some degree for a sufficiently narrow frequency band, if the frequency dependent complex gain that the digital filter needs to match (for cancellation) is not a sharply changing function of frequency in the narrow frequency band. If most of the unwanted noise below the earplug is in such a narrow frequency band where cancellation is possible, the ANR system can work well.

Periodic noise was used for the external sound field during the initial testing of the new DSP control system and ANREP's. Periodic noise is much easier to cancel. At low sound levels, the system performed very well. After altering the external sound field to band limited *random* noise (again at low sound levels) initial tests look very promising as shown in Figure 2.

Subsequent testing indicated that the frequency band where good cancellation occurred in Figure 2 could not be extended. For the two microphone feedforward ANREPs that were tested, not meeting the third criterion was the most detrimental problem. Let the gain of the digital filter be D. Define F as the transfer function of the speaker given by the output voltage of the error microphone divided by the voltage driving the speaker (in the absence of primary sound). Finally, let the passive attenuation of the earplug be represented by the transfer function P, defined by the output of the error microphone divided by the the input microphone (in the absence of secondary noise). For sound to be cancelled, $P + FD = 0$. The ANR system needs to do the opposite of "mother nature" or P. Since the speaker and microphone responses are relatively constant as a function of frequency, the digital filter gain needed for cancellation, $D = -P/F$, varies sharply as a function of frequency if P varies sharply as a function of frequency.

At frequencies where the earplug's passive control was significant ($>\approx$ 10 dB) the passive attenuation was a very sharply varying function of frequency for all of the earplugs that were tested. More than two dozen different types of earplugs were tested. Searching the literature and contacting experts on this issue did not help because surprisingly little is known about earplug passive attenuation data taken with a finer frequency resolution than every third octave band. Figure 3 shows the amplitude and phase of P from a typical earplug. This data was found by sine sweeping an external speaker and measuring P for various earplugs. In order for the feedforward approach to work well when cancelling random noise at all frequencies, the frequency dependent gain of the digital filter would have to compensate for these sharp variations in the passive attenuation of the earplug. For example, if the earplug's passive attenuation increases by 10 dB over a small frequency range and it is desirable for the system to cancel equally well at any frequency within this small frequency range, then the transfer function of the digital filter is required to to increase by 10 dB over a small frequency range.

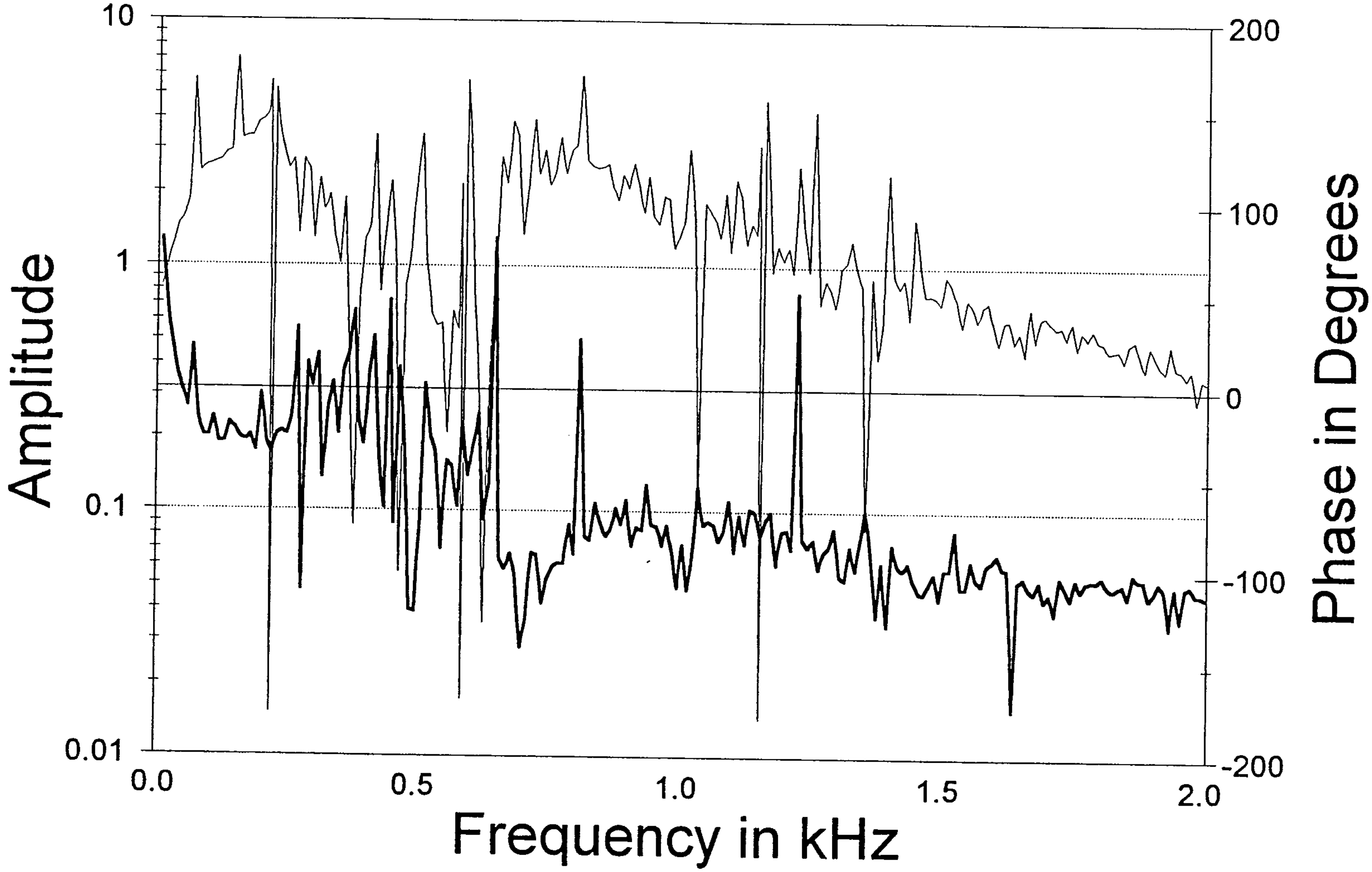
Because of the constraint of minimal delay, it is improbable that a causal digital filter



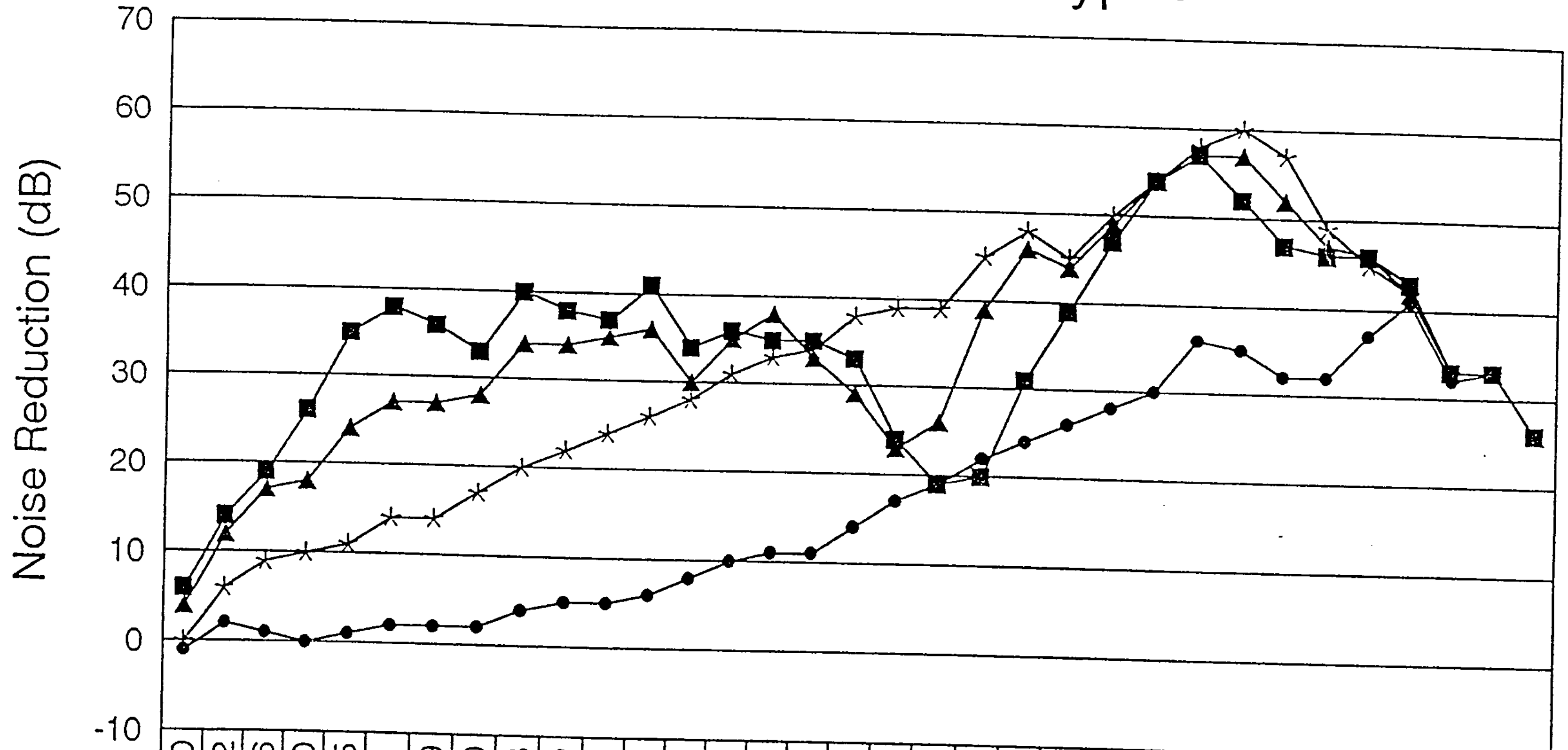
SPL at Kemar's Eardrum
Without and With Active Noise Control
Without: Total = 80.6 dB
With: Total = 61.1 dB
Magnitude (dB)
80
70
60
50
40
30
20
10
10^1 2 3 4 10^2 2 3 4 10^3 2 3 4 10^4 2
Frequency (Hz)

Figure 2.

Figure 3. P: Earplug Passive Control
(Error mic/Input mic)



Passive Attenuation Data
for all four ANREP Prototypes



Ear Plug:	10	12	16	20	25	31	39	50	63	79	99	125	157	198	250	315	397	500	630	794	1000	1260	1587	2000	2520	3175	4000	5040	6350	8000	10079	12699	16000
ANREP1 ●	-1	2	1	0	1	2	2	2	4	5	5	6	8	10	11	11	14	17	19	22	24	26	28	30	36	35	32	32	37	41	32	33	26
ANREP2 ✳	0	6	9	10	11	14	14	17	20	22	24	26	28	31	33	34	38	39	39	45	48	45	50	54	58	60	57	49	45	42	32	33	26
ANREP3 ▲	4	12	17	18	24	27	27	28	34	34	35	36	30	35	38	33	29	23	26	39	46	44	49	54	57	57	52	47	46	42	33	33	26
ANREP4 ■	6	14	19	26	35	38	36	33	40	38	37	41	34	36	35	35	33	24	19	20	31	39	47	54	57	52	47	46	46	43	33	33	26

Figure 4

exists which is well matched to a desirable complex gain with sharp frequency dependent variations. The output of a causal digital filter during the n^{th} iteration cannot depend on the input sampled at the $n+1$, $n+2$, ... iterations. Trying to define a digital filter that runs in real time and matches a desired frequency dependent complex gain is analogous to trying to expand an arbitrary function in a Fourier series that omits half of the normal modes.

Given additional time and effort, a different material might have been identified that could be made into an earplug with a complex passive transfer function that is both large in magnitude and is more constant as a function of frequency. Although we can not conclusively eliminate the feedforward approach as a viable option for making an optimal ANR earplug, our experiments found the feedforward approach to be problematic. Sharp frequency dependent variations in experimental measurements of P may have been caused by standing wave nodes, vibrations, or other spurious effects. For example, destructive interference may occur between primary sound transmitted through the earplug and non-human KEMAR "bone conduction".

It should be emphasized that the excellent results shown in Figure 2 were only found for ANREP1, which had exceptionally low passive control ($\leq$ 11 dB) in the frequency band where cancellation occurred (50 Hz to 315 Hz). Figure 4 gives the KEMAR measured passive control for the first four experimental NCPA ANR earplug prototypes. ANREP4 with only passive control would reduce the overall sound level better than ANREP1 with the addition of feedforward active control. Thus, an excessive sacrifice of potential passive control was needed to get the ANR results shown in Figure 2.

IV. The Feedback Algorithm

A schematic design of a digital ANR earplug using a feedback control loop is shown in Figure 5. Several feedback control algorithms were tested in conjunction with an ANR earplug. The algorithm finally chosen and incorporated into the negative feedback, adaptive digital control system is referred to as the C.O.D. algorithm (for Computationally Optimized Digital filter). This algorithm is described here.

In order to explain the feedback algorithms in more detail, a mathematical model of the system is now given.

Diagram 1. Mathematical Model of the System

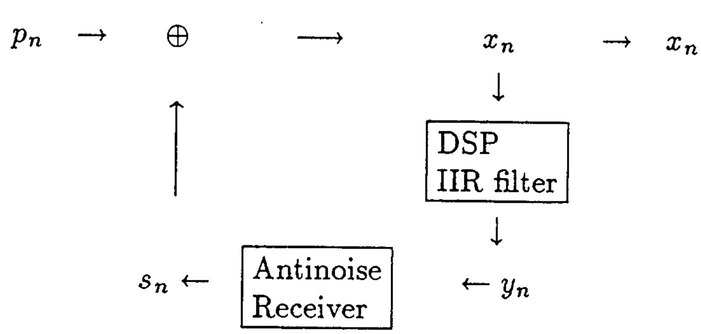
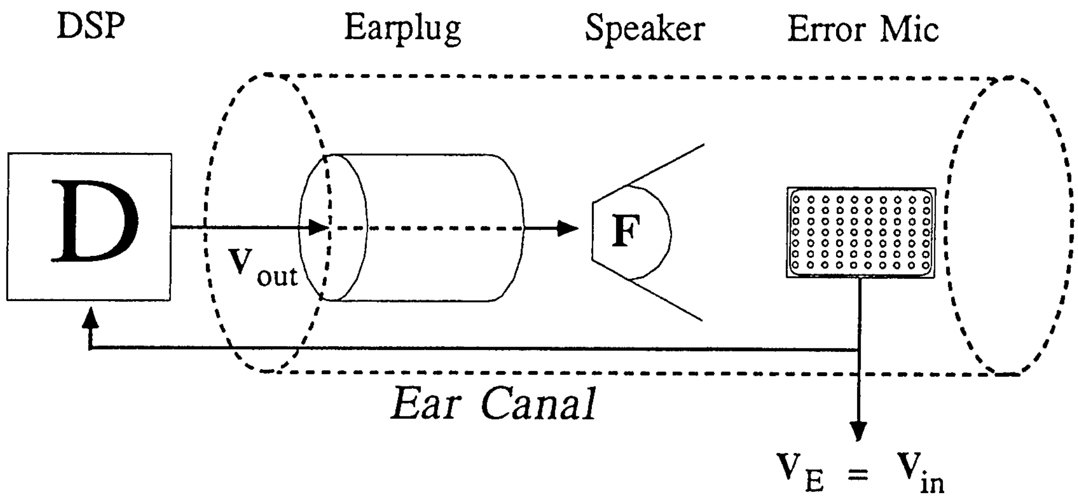


Figure 5. Schematic Depiction of ANR Earplug For Feedback Control Approach



In Diagram 1, the integer n advances by one after each iteration that x_n is input and y_n is output by the DSP. The sample period, τ, is the time required for each iteration. Here p_n represents the primary sound at the microphone, s_n represents the secondary sound at the microphone, and x_n represents the total noise at the microphone. $\{R_n\}_{n=1}^{m}$ represents the sampled response of the microphone to driving the speaker with a digital impulse voltage signal, given by $y_0 = 1$ and $y_n = 0$ for $n \neq 0$. The coefficients of the recursive IIR filter which the DSP implements are given by $\{a_k\}_{k=0}^{L}$ and $\{b_k\}_{k=1}^{J}$.

$$x_n = s_n + p_n \tag{1}$$

$$y_n = \sum_{k=0}^{L} a_k x_{n-k} + \sum_{k=1}^{J} b_k y_{n-k} \tag{2}$$

$$s_n = \sum_{k=1}^{m} R_k y_{n-k} \tag{3}$$

All of the feedback algorithms tested rely on knowledge of the response data, $\{R_k\}_{k=1}^{m}$. When the ANR system is initially turned on, no effort is made to cancel noise. Instead, the receiver is driven by a pulsed digital impulse, and each microphone response is recorded and subsequently averaged into a single set of response data. After performing such a calibration, the response data is known. Next, the acoustical power spectrum of the primary sound, $P_p(f)$, at the ANR microphone is measured by taking FFTs of digitally sampled data (modified with a Hanning window) and averaging the squared amplitudes.

For the COD algorithm, the coefficients that define the digital filter are chosen so as to minimize $\mathcal{L}$, the ear sensitivity weighted sound pressure level (SPL). Equations (4–5) define the mathematical model for $\mathcal{L}$.

$$\mathcal{L} = \int_0^\infty |T(f)|^2 P_p(f) 10^{[-M(f)+E(f)]} df \tag{4}$$

$$T(f) = \tag{5}$$

$$\frac{1 - \sum_{k=1}^{J} b_k \exp[2\pi i k f \tau]}{1 - \sum_{k=1}^{J} b_k \exp[2\pi i k f \tau] - \sum_{k=0}^{L} \sum_{j=1}^{m} a_k R_j \exp[2\pi i (k+j) f \tau]}$$

After numerically computing the optimal coefficients that define the IIR digital filter, the specified digital filter is run in real time to initiate ANR. The primary sound power spectrum is weighted by $M(f)$, in order to remove the frequency dependent sensitivity of the microphone, and $E(f)$, in order to introduce the frequency dependent sensitivity of the human ear. Equation (5) defines $T(f)$, an approximation of the transfer function of the residual sound (primary plus secondary) divided by the primary sound at the ANR microphone. The formula for $T(f)$ given here omits first order corrections as $f\tau$ becomes large and is only valid for frequencies well below the Nyquist frequency ($f\tau << 1/2$). In practice, it is typical for the large $f\tau$ breakdown in the $T(f)$ model too have little effect on $\mathcal{L}$ due to the diminishing value of $P_p(f)10 ** [E(f)]$.

After choosing a starting point in the digital filter parameter space, the COD algorithm iteratively calculates a trapezoidal approximation for the functional value, gradient and hessian of $\mathcal{L}$ and uses a modified Newton's method [29,30] to find a minimum. Applying the constraint of stability alone to the mathematical minimization problem can result in the design of a digital filter that makes the physical system feedback unstable. This can happen because there are differences between the computed response and the actual, current response of the physical system. In order to provide additional protection against instabilities it is desirable to set a minimal transient decay rate, d, which can be tolerated by the antinoise system. Thus the more stringent constraint of relative stability is imposed instead of an absolute stability. Each time $\mathcal{L}$ is evaluated, a check is made for insufficiently damped modes using the Routh-Hurwitz [31] technique. If such a mode exists, an extremely large "penalty" factor is added to $\mathcal{L}$. This prevents the minimization search from converging to a point in the digital filter parameter space where transients will (theoretically) linger longer than a maximal half-life $\gamma = \ln 2/d$. To be mathematically rigorous one would only apply Newton's method to unconstrained optimization problems. Further research might reveal a better choice of numerical optimization algorithm for this problem.

Advantages of the COD algorithm include: 1) The ability to increase the noise cancellation in the particular frequency bands where noise is perceived to be loudest to the human ear. 2) The automated mechanism for preventing feedback instabilities. Disadvantages include: 1) The complexity of the COD algorithm makes it cumbersome to implement. 2) Adapting for optimal control after changes in the unwanted noise currently requires pushing a button and waiting 15 to 30 seconds for the new digital filter to be calculated.

At the conclusion of the project, it was decided that the ANR performance would benefit from future efforts to make the mathematical model take into account the difference in sound levels at the error microphone and the eardrum. For the primary sound, define the transfer function $T_p(f)$ as the ratio of the sound at the eardrum divided by the sound at the ANR earplug microphone. In the presence of secondary sound, let $T_s(f)$ give the same ratio. Modifying the mathematical model to redefine $\mathcal{L}$ at the eardrum can be accomplished by redefining $T(f)$ in Eqn. (5).

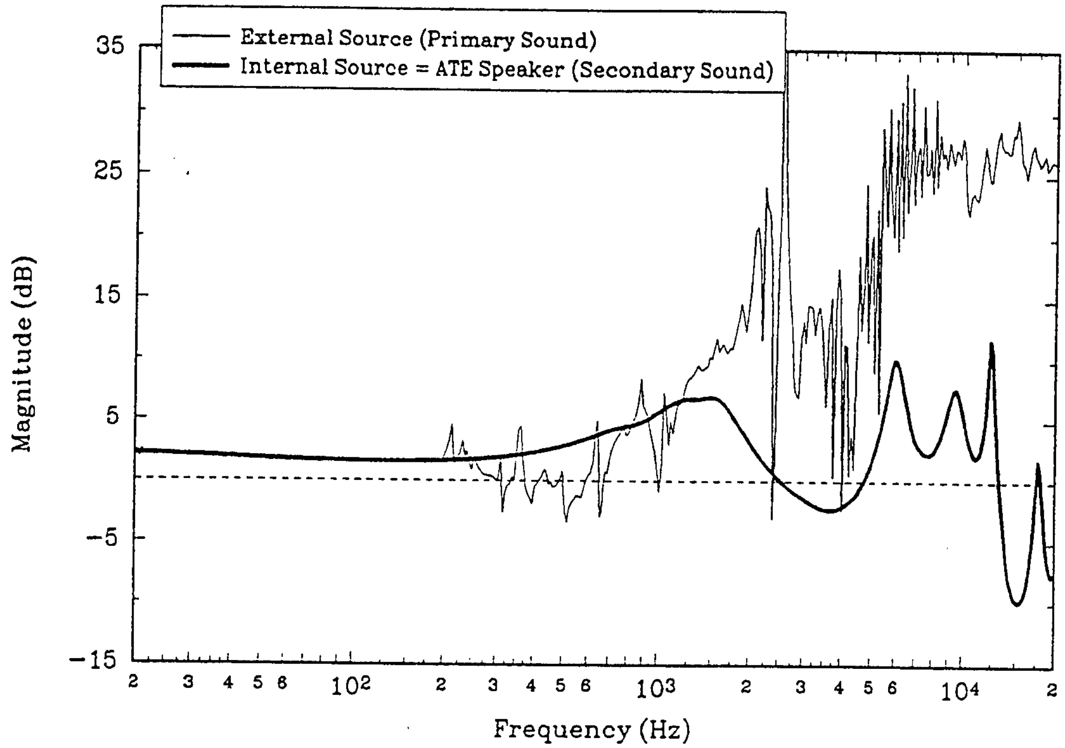
$$T(f) = \tag{6}$$

$$\frac{T_p(f)\left[1-\sum_{k=1}^{J} b_k \exp[2\pi i k f \tau]\right] + [T_s(f) - T_p(f)]\sum_{k=0}^{L}\sum_{j=1}^{m} a_k R_j \exp[2\pi i (k+j) f\tau]}{1-\sum_{k=1}^{J} b_k \exp[2\pi i k f\tau] - \sum_{k=0}^{L}\sum_{j=1}^{m} a_k R_j \exp[2\pi i (k+j) f\tau]}$$

A second related correction would involve refining the ear sensitivity weighting. The A-weighting, for example, is defined relative to SPLs measured at the opening of the ear when no obstructing device is fit into the ear canal. The ratio of eardrum to above the ear SPLs shows a resonance in humans that is typically between 2 and 4 kHz. This resonance effect is reflected in the A-weighting, which can be thought of as the combination of ear drum sensitivity and the ear canal acoustics that determine the transfer function from above the ear to ear drum SPLs. Having calculated the sound at the eardrum, however, one is only interested in the ear drum sensitivity.

Figure 6.

Eardrum/ATE Knowles microphone Frequency Response data from Anechoic Room using various sound sources.



Eardrum/ATE Knowles microphone Phase data from Anechoic Room using various sound sources.

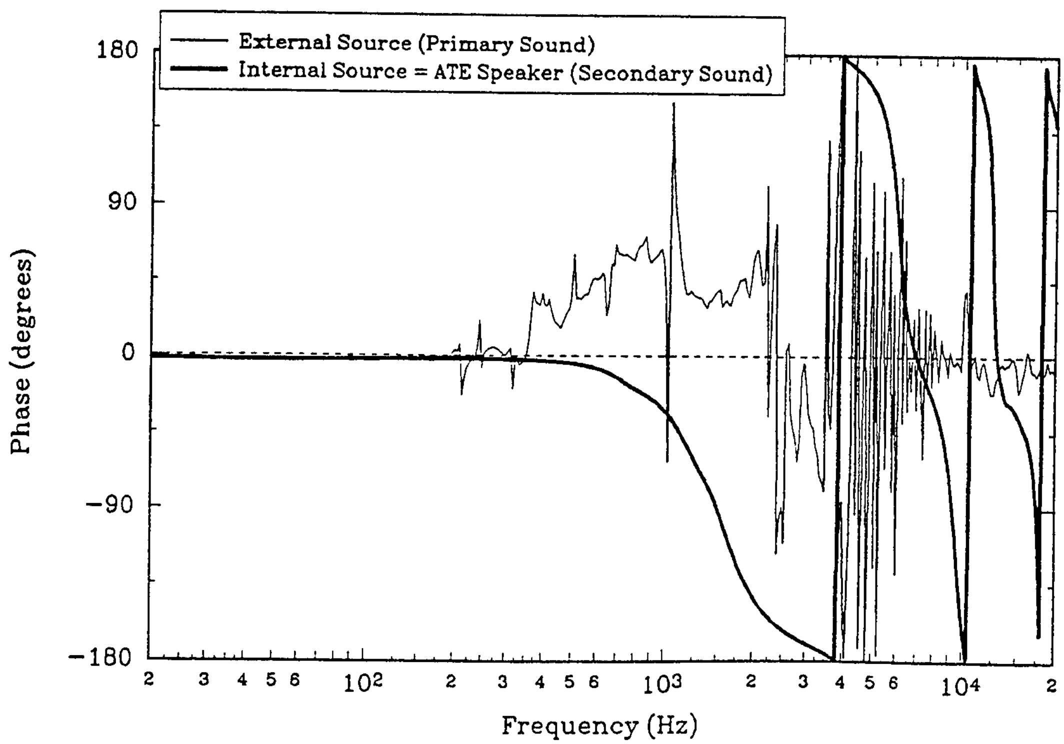
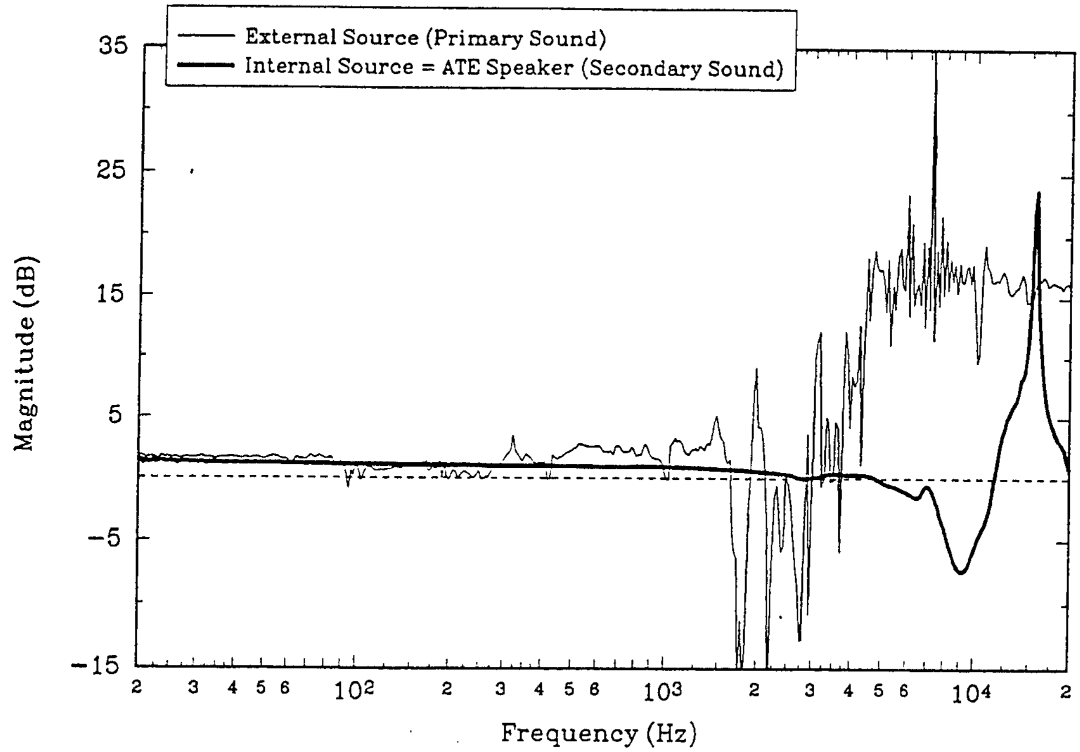
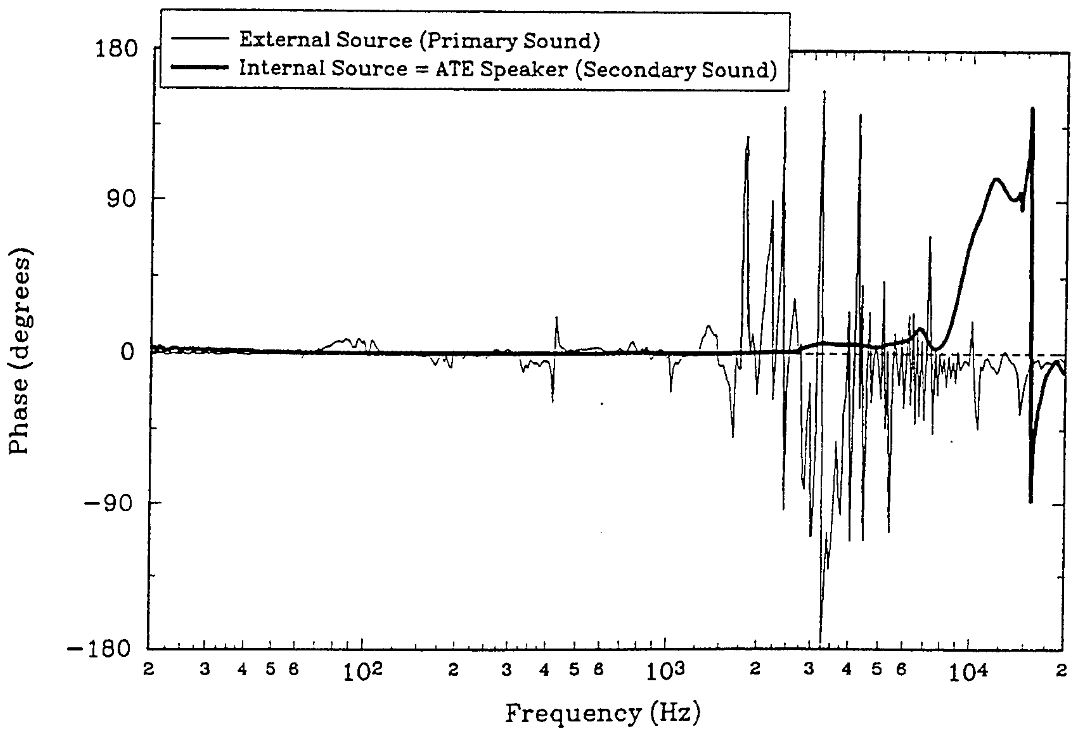


Figure 7. Eardrum/ITE Knowles microphone Frequency Response data from Anechoic Room using various sound sources.



Eardrum/ITE Knowles microphone Phase data from Anechoic Room using various sound sources.



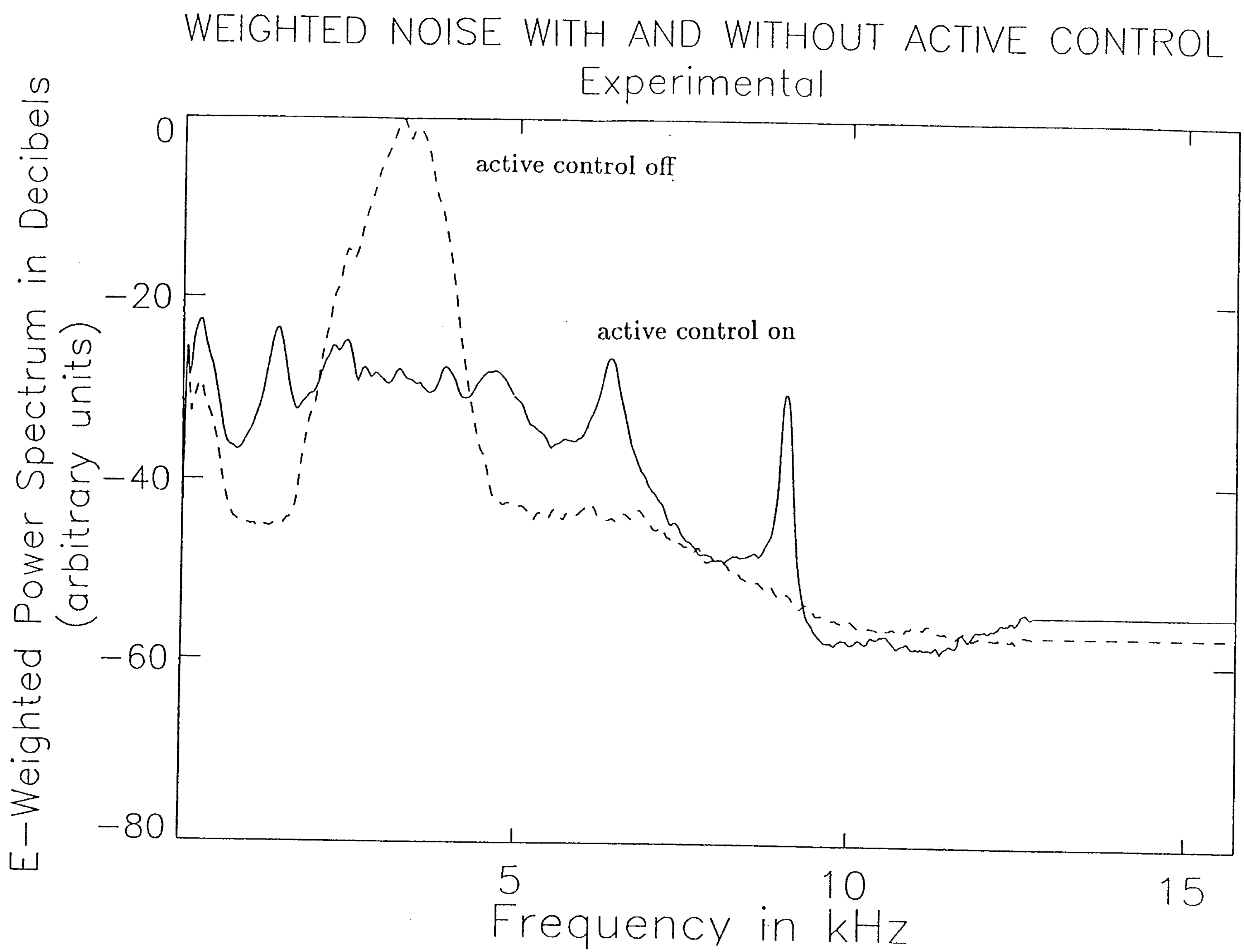

WEIGHTED NOISE WITH AND WITHOUT ACTIVE CONTROL
Experimental
E—Weighted Power Spectrum in Decibels
(arbitrary units)
0
−20
−40
−60
−80
active control off
active control on
5
10
15
Frequency in kHz

Figure 8.

Figures 6 and 7 illustrate $T_s(f)$ (dark curves) and $T_p(f)$ (light curves). A future correction of the mathematical model would take into account the fact that $T_s(f) \neq T_p(f) \neq 1$. (The $T_s(f)$ curve in Figure 6 reflects the fact that the ATE device essentially lengthens the KEMAR ear canal, which reduces the first resonant frequency to below 2 kHz.) These graphs were taken in the NCPA anechoic room, where it was difficult to generate enough sound to penetrate a headset and earplug in order to measure $T_p(f)$ accurately. Another minor problem is that the two EK-3132 microphones appear to need different calibration corrections, since their ratio is offset from 0 dB in the low frequency limit. The data is adequate to indicate that $T_s(f) \neq T_p(f) \neq 1$ for the ATE device and the ITE results show relatively less disagreement between the two microphones.

Figure 8 shows the ANR *microphone* results obtained using a similar control system with an ANR headset consisting of a Sony MDR-CD6 speaker and Knowles BT-1759 microphone. This illustrates the potential of ANR performance that can be obtained at the *eardrum* if $T_p(f)$ and $T_s(f)$ are known.

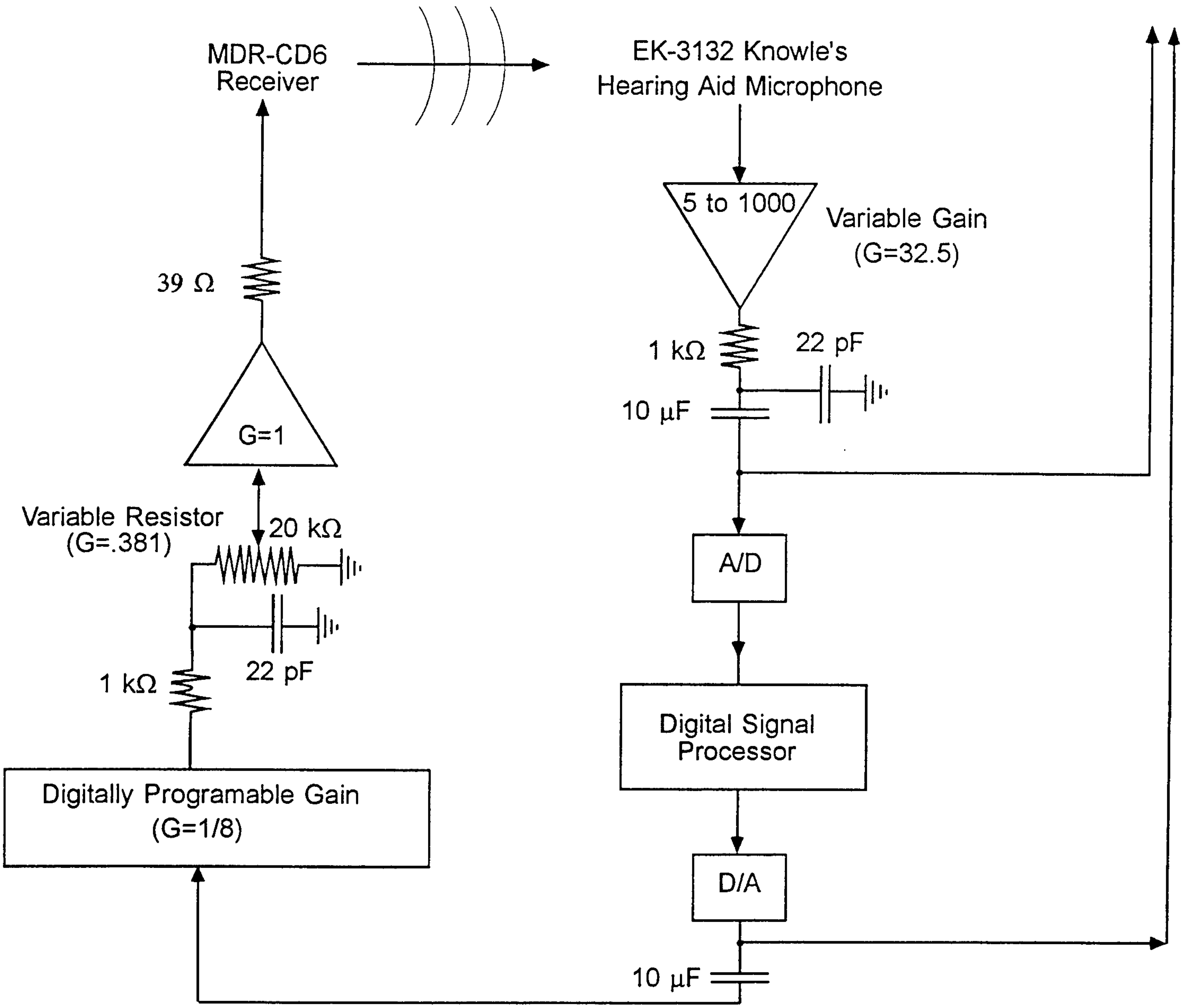
V. Specifications for Components of the ANR Earplugs

The digital filter defined by the technique described in the previous section is incorporated into an electroacoustical negative feedback loop, which is depicted in Figure 9. Antialiasing and reconstruction filters are made from RC filters. The presence of both filters reduces the SPL of the acoustical feedback by 6 dB at 7 kHz and approximately another 12 dB for each additional octave. Typically the digital filter is run at a sample rate of about 20 kHz, giving a Nyquist frequency of 10 kHz. Filters which cutoff at lower frequencies and roll-off more sharply have the drawback of increasing the time delay of the feedback response.

Figure 10 depicts the two ANR earplug systems that were developed. The in the ear (ITE) ANR earplug has superior passive and active control, but in its current embodiment it is not fully human compatible. A future research and development effort could make the ITE device fully human compatible, but for the current project it was a goal to develop an ITE prototype that demonstrated the principle and potential. The speaker and microphone have relatively large exposed metallic parts that make it incompatible with human ears, especially for people with relatively small ears. The above the ear (ATE) earplug is fully human compatible. The ATE speaker uses a Sony MDR-CD6 speaker and a Knowles EK-3132 hearing aid microphone. The ITE device uses the same type of microphone and a custom made piezoelectric transducer.

The causality issue discussed before also applies to feedback control systems. The digital filter is limited in its capability to approximate a desirable frequency dependent gain. It is optimal for the microphone, speaker, and entire feedback loop to have a relatively "flat" (constant in amplitude and phase) response as a function of frequency. Since the causal digital filter is limited in how well it can compensate for resonances in the ANR speaker, it is advantageous to have ANR speakers with a very flat response.

Figure 11 gives the amplitude of the frequency response of the ATE and ITE ANR earplugs speakers as measured at the KEMAR ear drum. The piezoelectric device is extremely flat. The ATE device has a problematic resonance from 1 to 2 kHz. Acoustically



MDR-CD6
Receiver
EK-3132 Knowle's
Hearing Aid Microphone
5 to 1000
Variable Gain
(G=32.5)
39 Ω
1 kΩ
22 pF
10 μF
G=1
Variable Resistor
(G=.381)
20 kΩ
A/D
22 pF
1 kΩ
Digital Signal
Processor
Digitally Programable Gain
(G=1/8)
D/A
10 μF

Figure 9.

Figure 10.
Active Noise Reduction Earplug Configurations

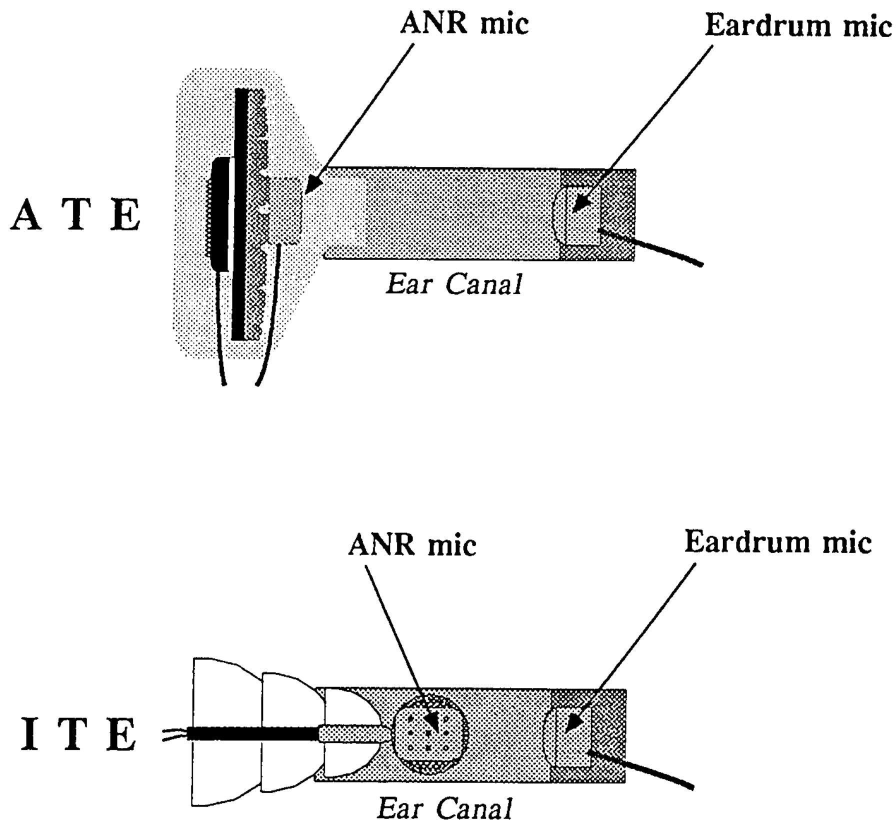
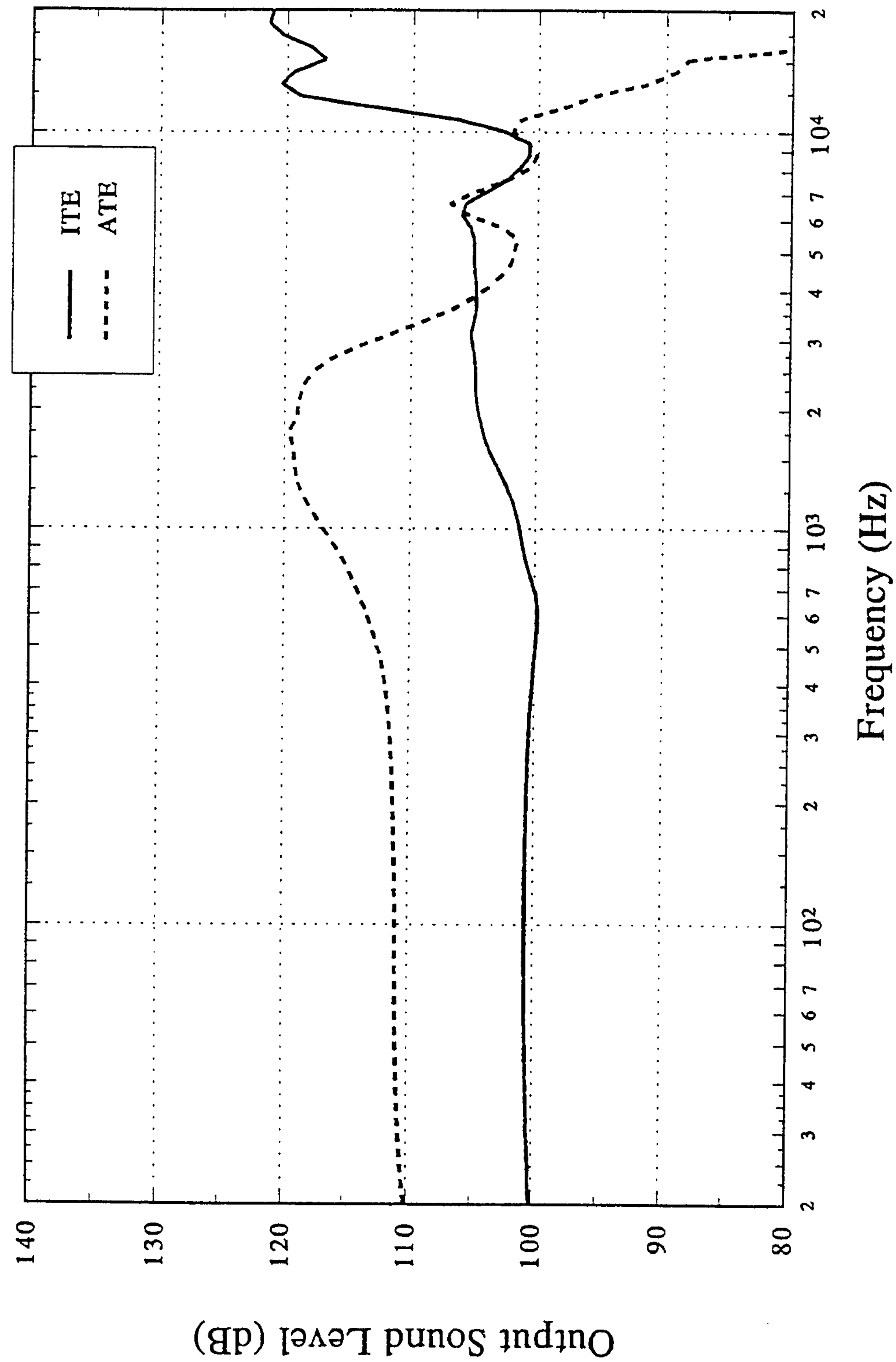


Figure 11.

Frequency Response of ITE and ATE

Driving Voltage for ITE @ 10 V_{rms} and ATE @ .2 V_{rms}



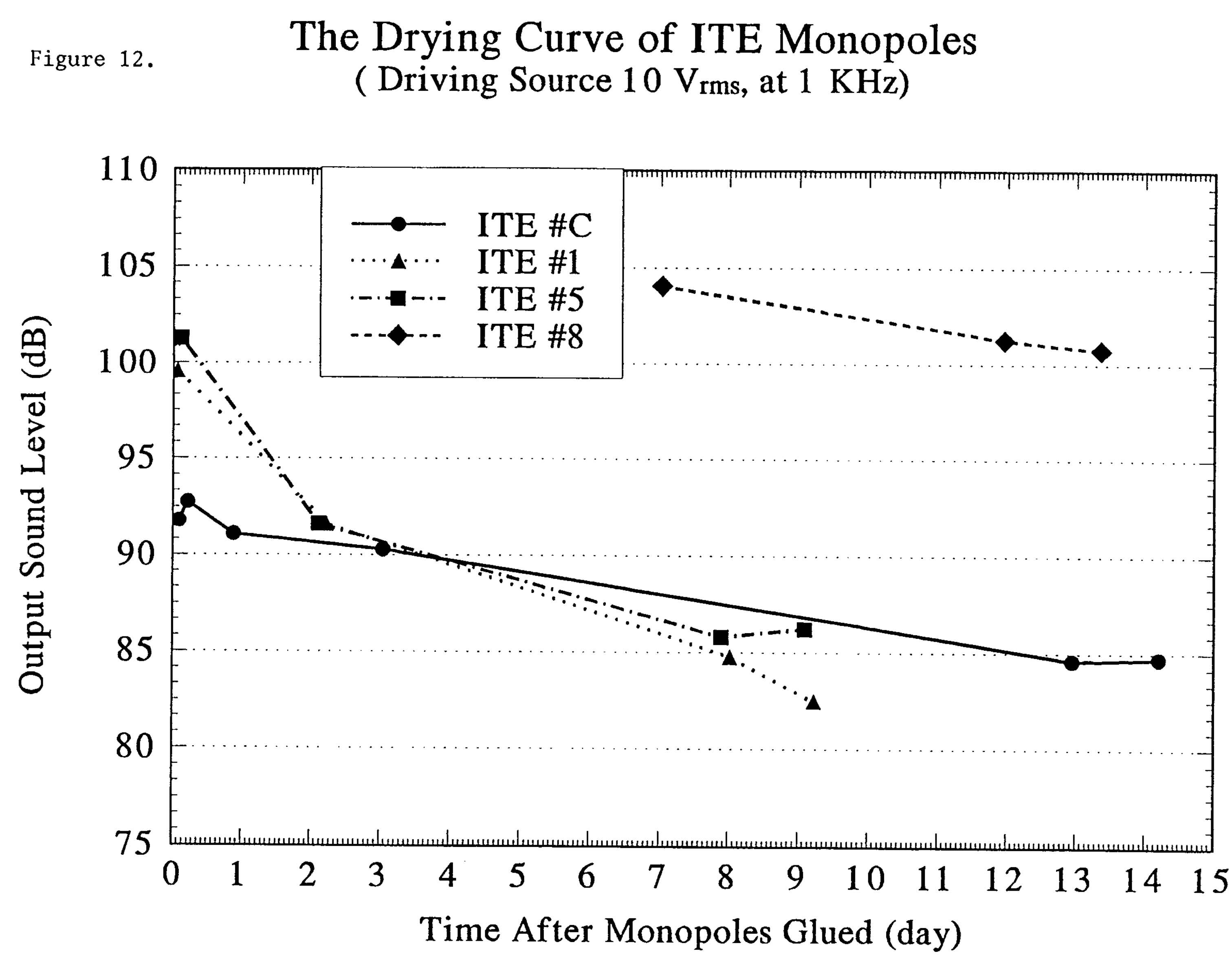
absorbing materials in front of the Sony MDR-CD6 speaker in the ATE device reduce the amplitude of this resonance. In practice, the performance of the ATE device was limited by instabilities between 1 and 2 kHz. An ANR headset made at the NCPA with the same speaker and a similar Knowles microphone have achieved better active control results than the ATE ANR earplug. The resonance seems to be symptomatic of the ATE's geometry. Further damping of the resonance appeared to excessively increase the response delay time. Further effort to improve the ATE design may resolve this resonance problem.

For active noise reduction, the magnitude of secondary noise must be equal to the primary noise. This becomes an important consideration at 140 dB SPL. It is not difficult to reach sufficiently loud sound levels that exist under a headset and earplug using the Sony MDR-CD6 of the ATE ANR earplug. For the custom made ITE piezoelectric speaker, reaching sufficiently loud sound levels was more challenging. A limit on the driving voltage of 10 V (as in Fig. 11) was chosen for practical reasons, since additional amplifiers would have to be incorporated into the circuitry if the $\pm$ 10 V dynamic range output by the D/A converters was insufficient. Also, the necessity of higher driving voltages might constitute another hurdle in the path of making the device safe for human use. It was discovered that the glue chosen to clamp the perimeter of the speaker "cone" had a large effect. Apparently, as the glue dried the clamping increased and the acoustical output of the miniature speaker decreased. Figure 12 shows data from the final ITE design chosen (ITE #8) as well as prior ITE speaker prototypes.

The design of the digital control system assumes that the entire negative feedback control loop is linear. It is important, therefore, for the ANR speaker and microphone to be operated in a linear regime. Raising the external SPL to 140 dB causes previously existing ANR devices to be extended well beyond their linear regime of operation, resulting in failure. The ANR microphone incorporated into both the ATE and ITE ANR earplug is a Knowles EK-3132. This microphone, with a 10 V bias voltage, begins to go non-linear at about 107 dB. Figure 13 shows the linearity of the ATE and final ITE (#8) speaker. The y-axis shows how much the sound increased when the 1 kHz driving voltage was doubled. For a perfectly linear speaker, the SPL should increase by 6.02 dB. The x-axis gives the SPL at the higher of the two sound levels (after doubling the voltage).

From the time domain perspective it is desirable for an ANR speaker to respond as quickly as possible. From the frequency domain perspective, the phase of the frequency response (as measured by the microphone of the feedback loop) should change as slowly as possible as a function of frequency. The time delay required for a speaker to respond is reflected by the tendency for the phase of the speaker's frequency response to increase linearly with frequency. If it were possible for the phase delay to tend to *decrease* linearly as a function of frequency this would be good, since this would compensate for the effect of time delays in the rest of the feedback loop. In practice, however, the speakers typically make a large contribution to the time delay of the feedback loop. After the D/A converter (see Fig. 9) starts to output a digital impulse lasting 50 usec, the peak response seen on the filtered microphone output just in front of the A/D converter occurs 73 usec later. Under the same conditions, the ATE peak response occurs 170 usec later. When the ANR control system is running, approximately 5 usec more time is required for the digital filter system (A/D, DSP, D/A) to complete the feedback loop.

Figure 12.



The Drying Curve of ITE Monopoles
(Driving Source 10 Vrms, at 1 KHz)
Output Sound Level (dB)
110
105
100
95
90
85
80
75
0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15
Time After Monopoles Glued (day)
ITE #C
ITE #1
ITE #5
ITE #8

Figure 13. Linearity of ITE Monopoles and ATE # 6
(Driving Source: 1 KHz Fixed Sine Wave)

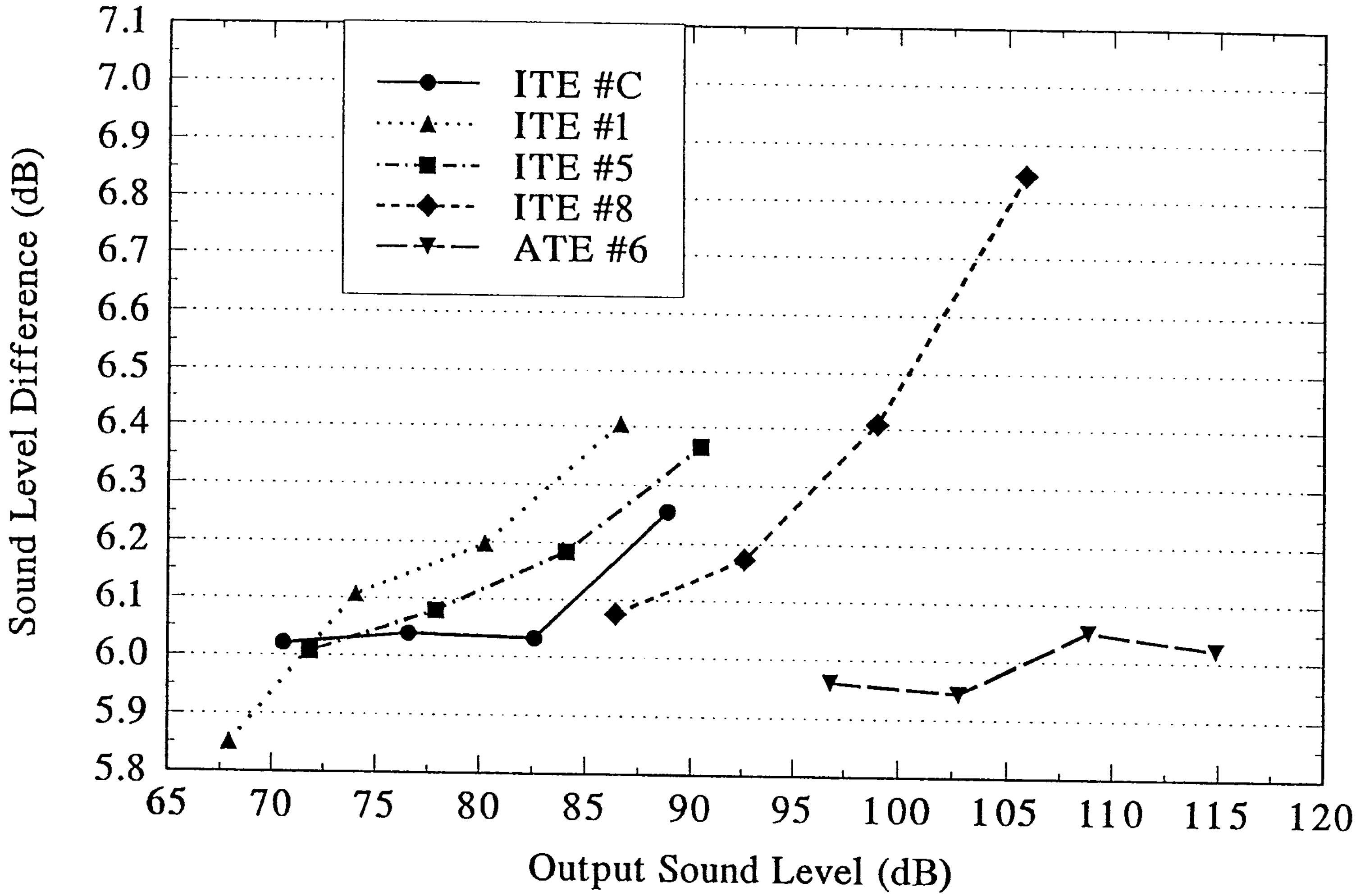


Figure 14. Normalized Autocorrelation Curve of ATE

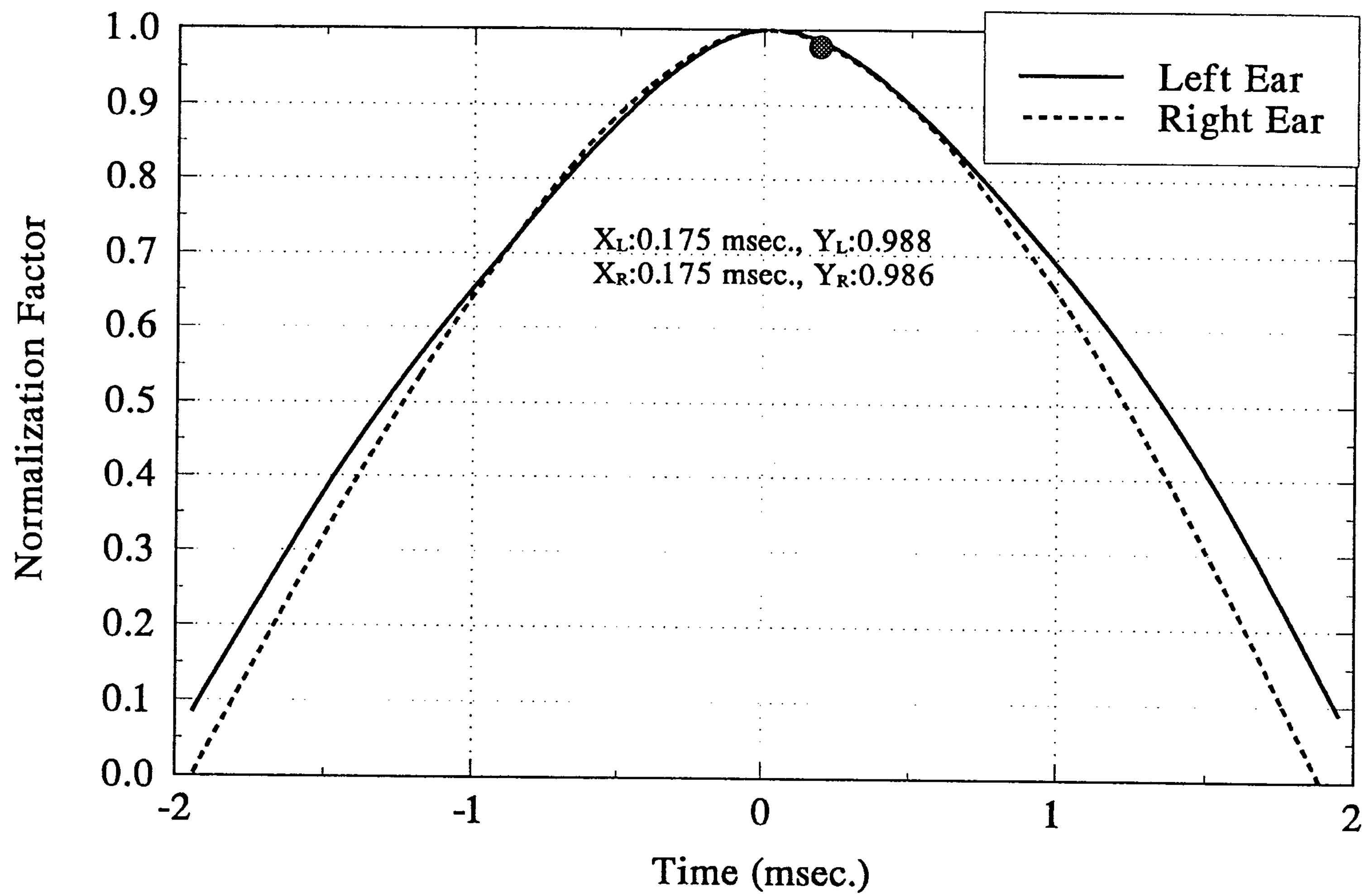
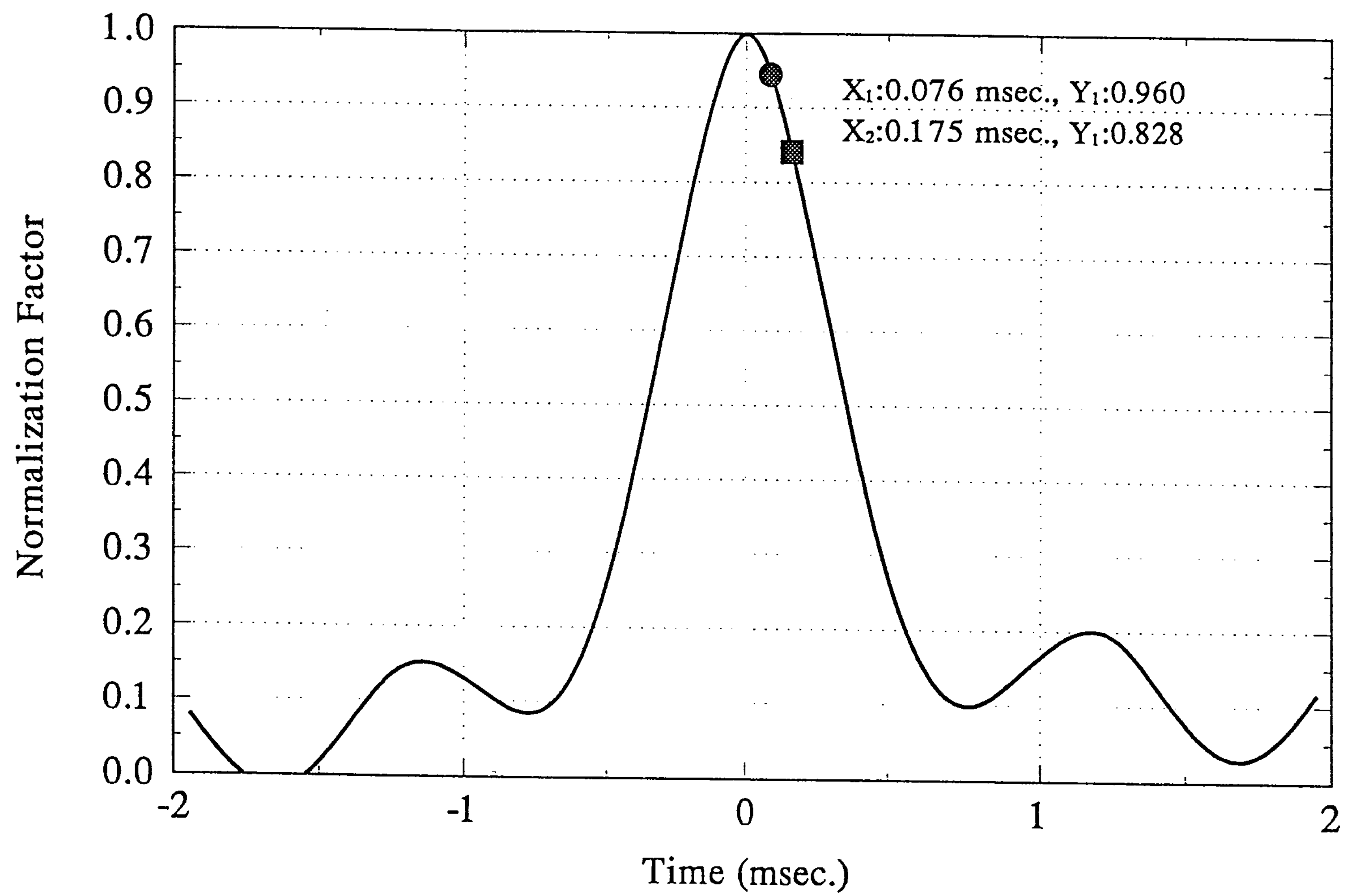


Figure 15. Normalized Autocorrelation Curve of ITE



To illustrate the advantage of having a fast response, it is useful to examine the autocorrelation of the noise each system is trying to cancel. For the ATE device under 140 dB of jet noise, the autocorrelation of primary sound in the KEMAR mannequin is given by Figure 14. Figure 15 gives the same data for the ITE device. Assume that the ATE device were used in the same primary sound field given in Figure 15. The ATE system can at best detect noise and begin to cancel 175 usec later. But after 175 usec, only 83 percent of the detected primary sound is still present. Although the feedback control system has many other limiting factors, this factor alone places an upper limit on overall noise reduction at 83 percent or 15 dB.

From the frequency domain perspective, the relatively higher passive attenuation of the ITE device at low frequencies typically results in sound below the earplug that has a less band limited spectrum with relatively more of the total acoustical power at higher frequencies. This type of spectrum is more difficult to cancel. This same information can be viewed from the time domain perspective. Relative to the ATE device (Figure 14) the autocorrelation of the sound under the ITE device (Figure 15) drops off more sharply as a function of the time delay.

VI. Performance of the ANR Earplugs

A. National Center for Physical Acoustics Experiments

Figure 16 through 18 show an experimental run for the ATE system, which includes a 9AN/2 headset. In order to insure that the ITE device is operated in a linear regime, the overall SPL of the simulated jet noise in the NCPA noise reverberation room was reduced from 140 to 125 dB, as Figure 19 depicts. Figure 20 shows the noise spectrum at the KEMAR ear drum before and after the ITE ANR is turned on. Note that matching the 85 dB SPL of primary sound is well with in the linear regime of the ITE #8 speaker (see Fig. 13). The ITE cancellation band goes from 16 to 3175 Hz.

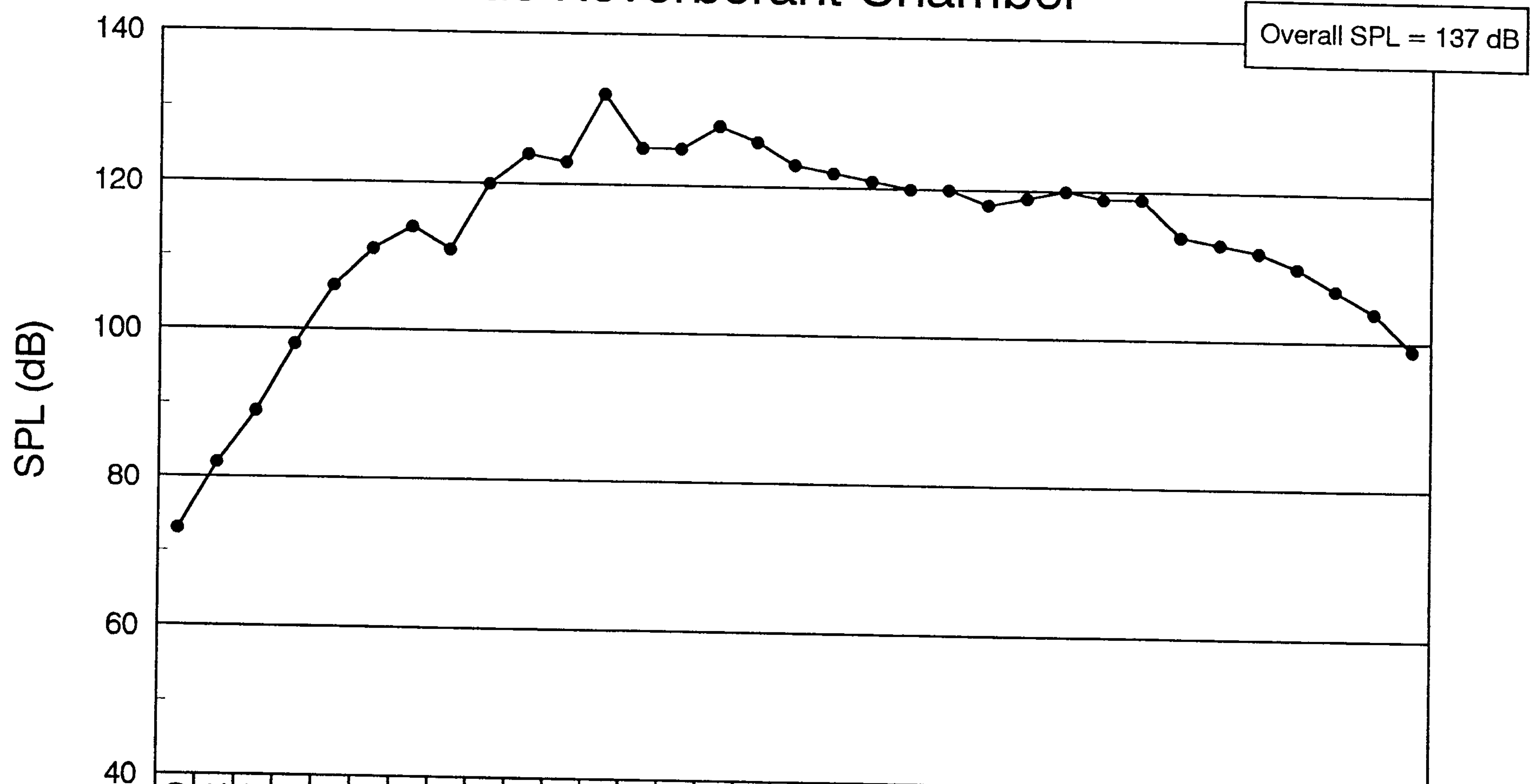
Figure 21 shows some disagreement between the amount of active noise reduction measured at the ANR microphone and the KEMAR ear drum microphone. The disagreement is relatively high at the adjacent third octave bands centered at 397, 500 and 630 Hz. The ATE disagreement (in Fig. 18) is worse than the ITE, and 500 and 630 Hz are again relatively problematic. Much more disagreement was measured when using 1/2" B&K microphones at the KEMAR ear drum. In addition to the problem with vibrations, some disagreement between the two microphones was observed to be correlated with standing wave modes in the NCPA reverberation room. If a standing wave node is close to the two microphones, then there can be a significant difference in the SPL at the two microphones. Removing the standing waves is not a fully satisfactory solution here, since hush houses and many other noisy environments also have standing waves. In addition, standing wave modes do not appear to fully account for these low frequency discrepancies.

The overall SPL reductions were 11.8 dB for the ITE, and 9.3 dB for the ATE ANR earplug. Use of the C.O.D. algorithm would not be justified if the numerically computed theoretical transfer function (at the ANR microphone) given by Eqn. (7) does not match experimental results. Figures 22 and 23 compare theory and experiment. Here a negative

Figure 16.

Absolute Sound Levels

Inside Reverberant Chamber

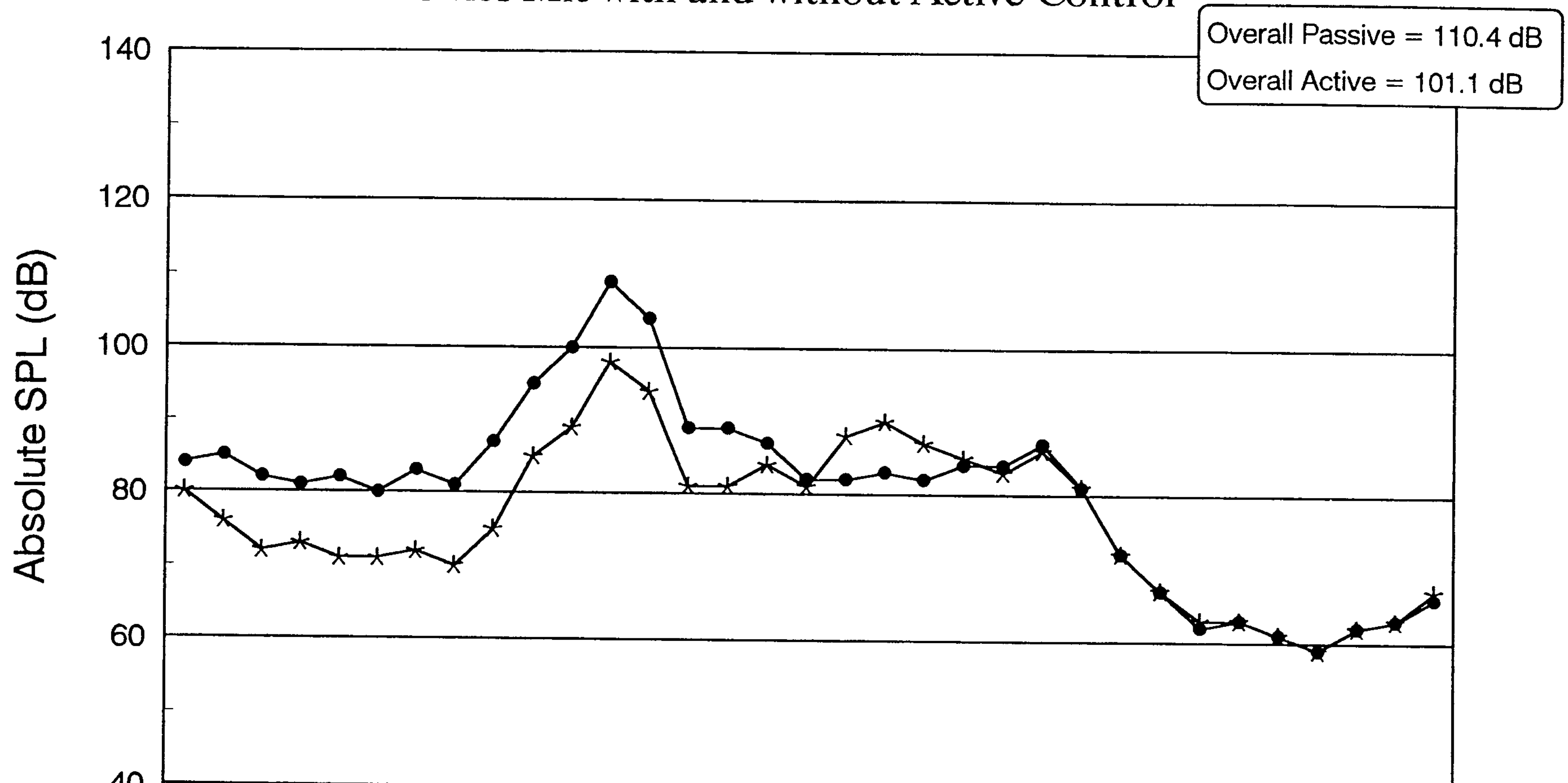


	10	12	16	20	25	31	39	50	63	79	99	125	157	198	250	315	397	500	630	794	1000	1260	1587	2000	2520	3175	4000	5040	6350	8000	10079	12699	16000
B&K Mic	73	82	89	98	106	111	114	111	120	124	123	132	125	125	128	126	123	122	121	120	120	118	119	120	119	119	114	113	112	110	107	104	99

Figure 17.

ATE Spectrum of Right Eardrum

Knowles Mic with and without Active Control

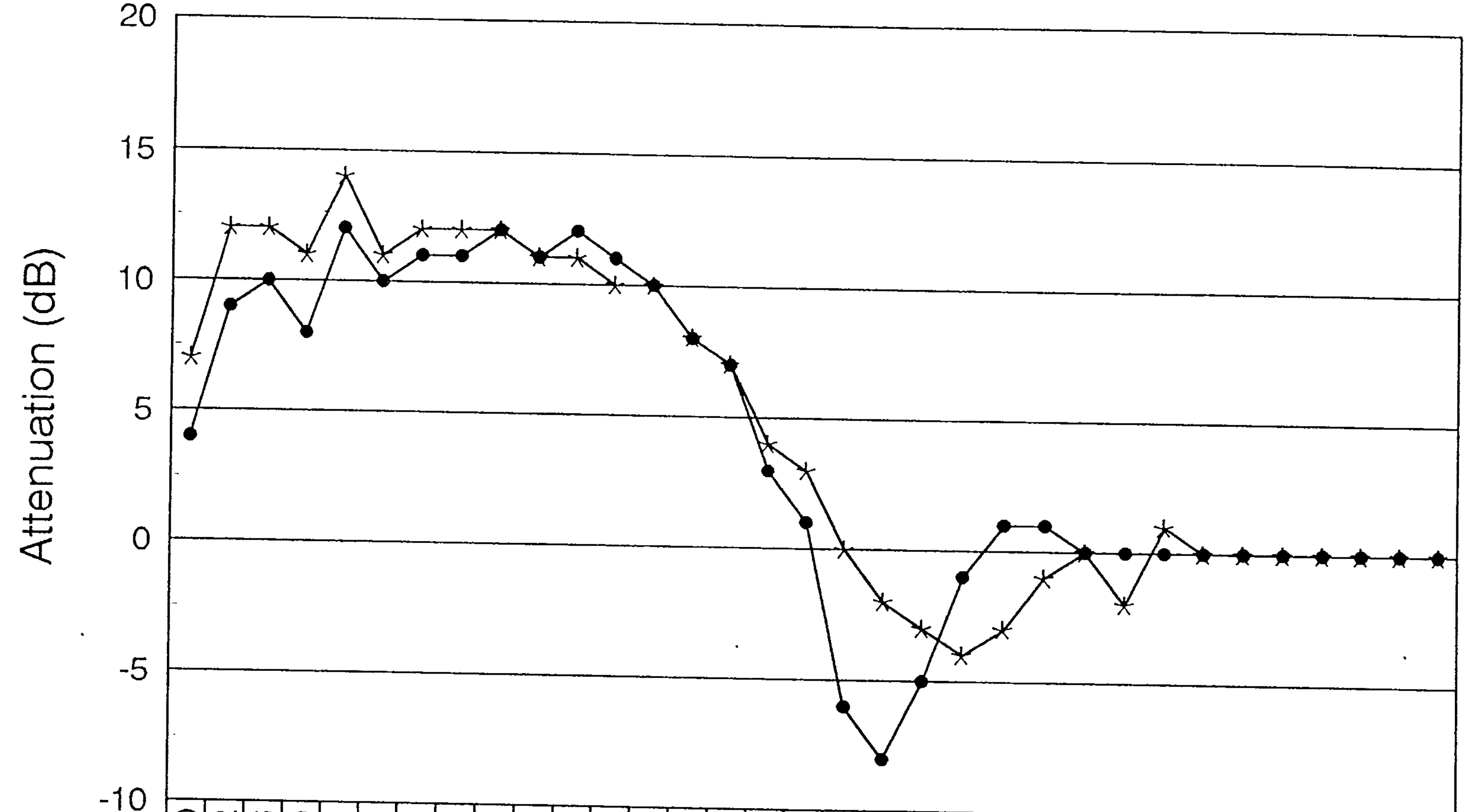


	10	12	16	20	25	31	39	50	63	79	99	125	157	198	250	315	397	500	630	794	1000	1260	1587	2000	2520	3175	4000	5040	6350	8000	10079	12699	16000
Passive ●	84	85	82	81	82	80	83	81	87	95	100	109	104	89	89	87	82	82	83	82	84	84	87	81	72	67	62	63	61	59	62	63	66
Active ✶	80	76	72	73	71	71	72	70	75	85	89	98	94	81	81	84	81	88	90	87	85	83	86	81	72	67	63	63	61	59	62	63	67

Figure 18.

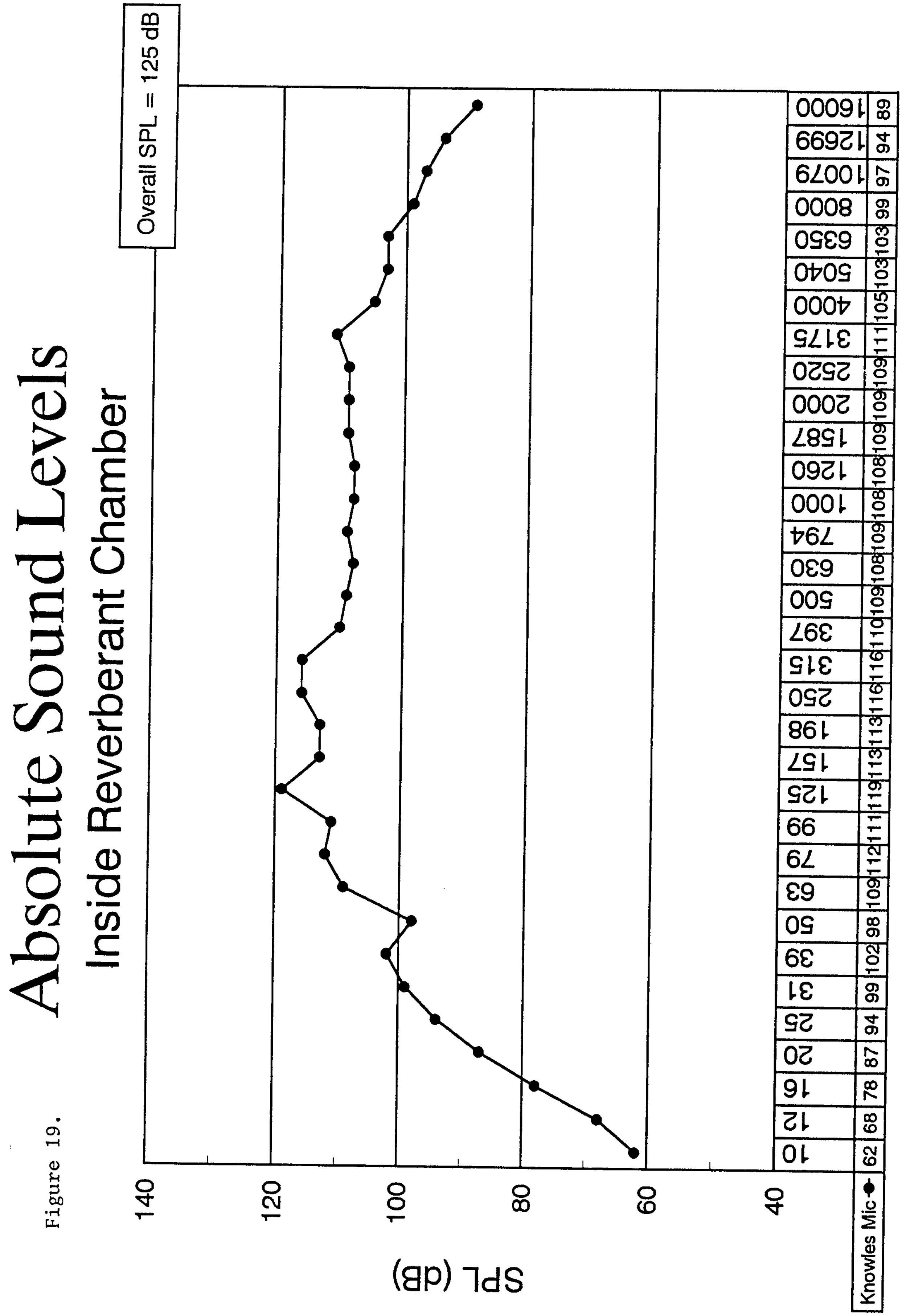
Right ear... ANR using ATE

(Knowles Eardrum Mic vs. ATE Knowles Mic)



	10	12	16	20	25	31	39	50	63	79	99	125	157	198	250	315	397	500	630	794	1000	1260	1587	2000	2520	3175	4000	5040	6350	8000	10079	12699	16000
Knowles Eardrum ●	4	9	10	8	12	10	11	11	12	11	12	11	10	8	7	3	1	-6	-8	-5	-1	1	1	0	0	0	0	0	0	0	0	0	0
ATE Knowles Mic ✳	7	12	12	11	14	11	12	12	12	11	11	10	10	8	7	4	3	0	-2	-3	-4	-3	-1	0	-2	1	0	0	0	0	0	0	0

Figure 19.

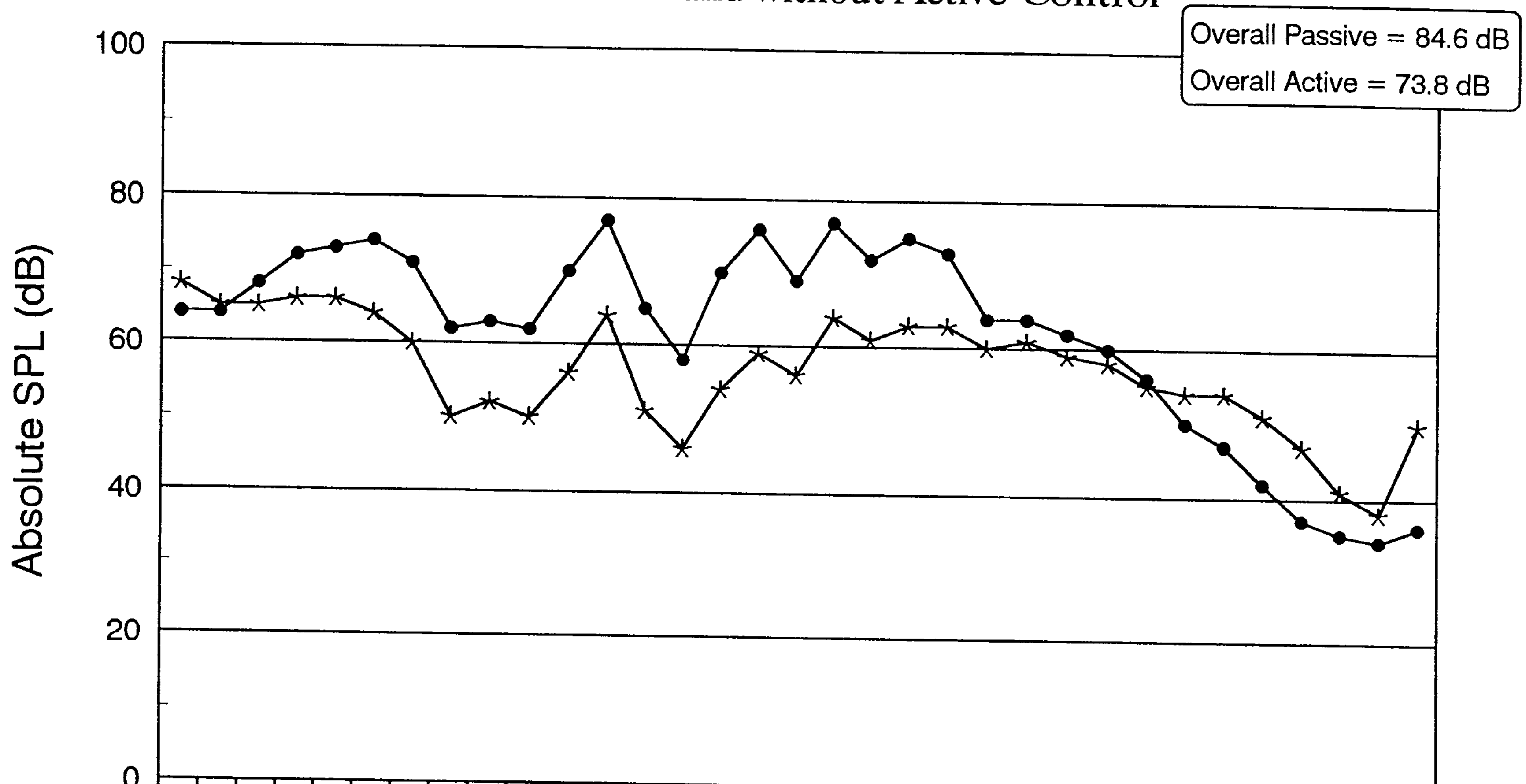


Frequency	Knowles Mic
10	62
12	68
16	78
20	87
25	94
31	99
39	102
50	98
63	109
79	112
99	111
125	119
157	113
198	113
250	116
315	116
397	110
500	109
630	108
794	109
1000	108
1260	108
1587	109
2000	109
2520	109
3175	111
4000	105
5040	103
6350	103
8000	99
10079	97
12699	94
16000	89

Figure 20.

ITE Spectrum of Left Eardrum

Knowles Mic with and without Active Control



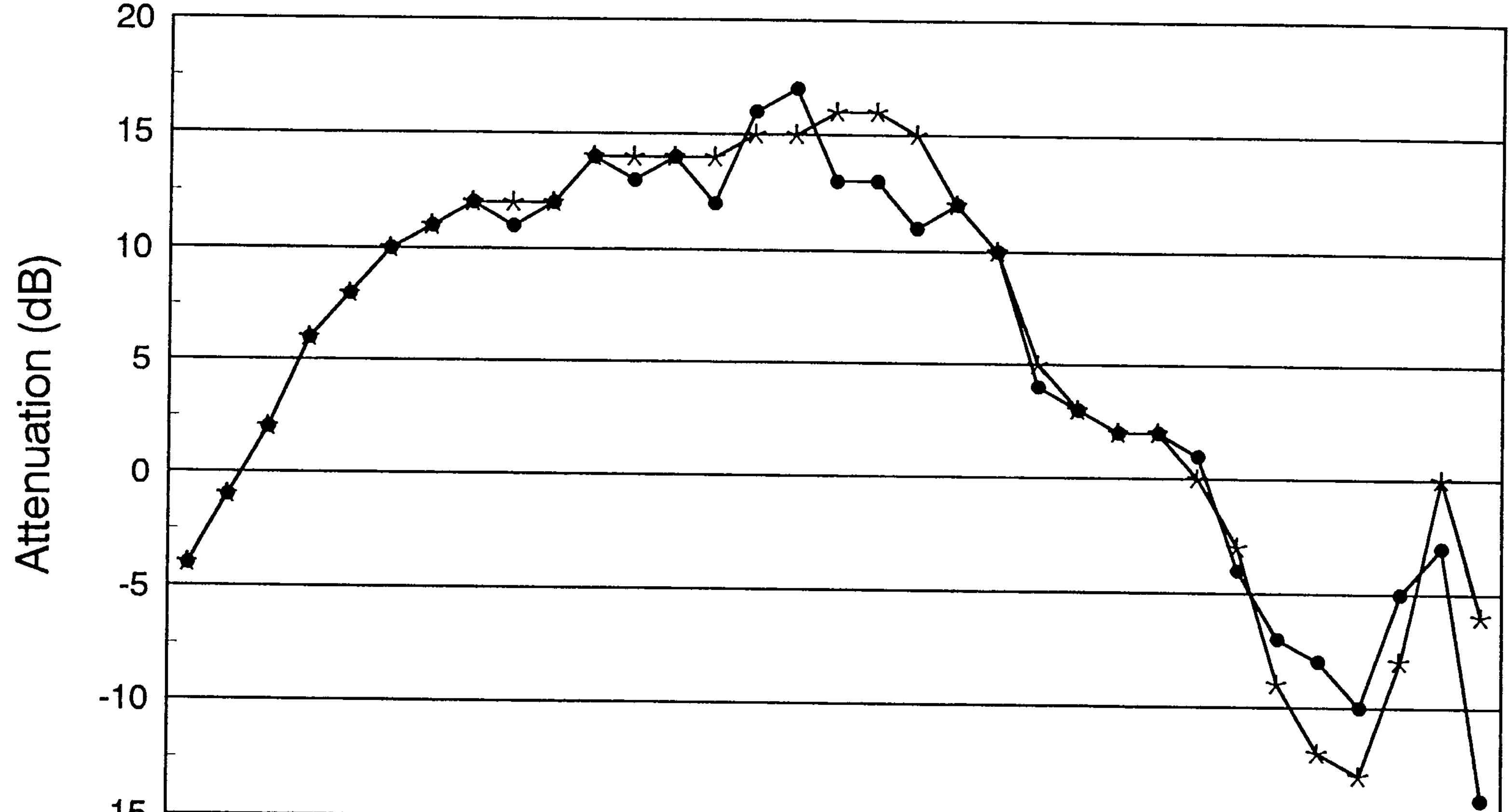
	10	12	16	20	25	31	39	50	63	79	99	125	157	198	250	315
Passive ●	64	64	68	72	73	74	71	62	63	62	70	77	65	58	70	76
Active ✳	68	65	65	66	66	64	60	50	52	50	56	64	51	46	54	59

	397	500	630	794	1000	1260	1587	2000	2520	3175	4000	5040	6350	8000	10079	12699	16000
Passive ●	69	77	72	75	73	64	64	62	60	56	50	47	42	37	35	34	36
Active ✳	56	64	61	63	63	60	61	59	58	55	54	54	51	47	41	38	50

Figure 21.

Left ear... ANR using ITE

(Knowles Eardrum Mic vs. ITE Knowles Mic)



	10	12	16	20	25	31	39	50	63	79	99	125	157	198	250	315	397	500	630	794	1000	1260	1587	2000	2520	3175	4000	5040	6350	8000	10079	12699	16000
Knowles Eardrum ●	-4	-1	2	6	8	10	11	12	11	12	14	13	14	12	16	17	13	13	11	12	10	4	3	2	2	1	-4	-7	-8	-10	-5	-3	-14
ITE Knowles Mic ✱	-4	-1	2	6	8	10	11	12	12	12	14	14	14	14	15	15	16	16	15	12	10	5	3	2	2	0	-3	-9	-12	-13	-8	0	-6

Figure 22. Comparison of Calculated and Experimental Cancellation
ATE right ear

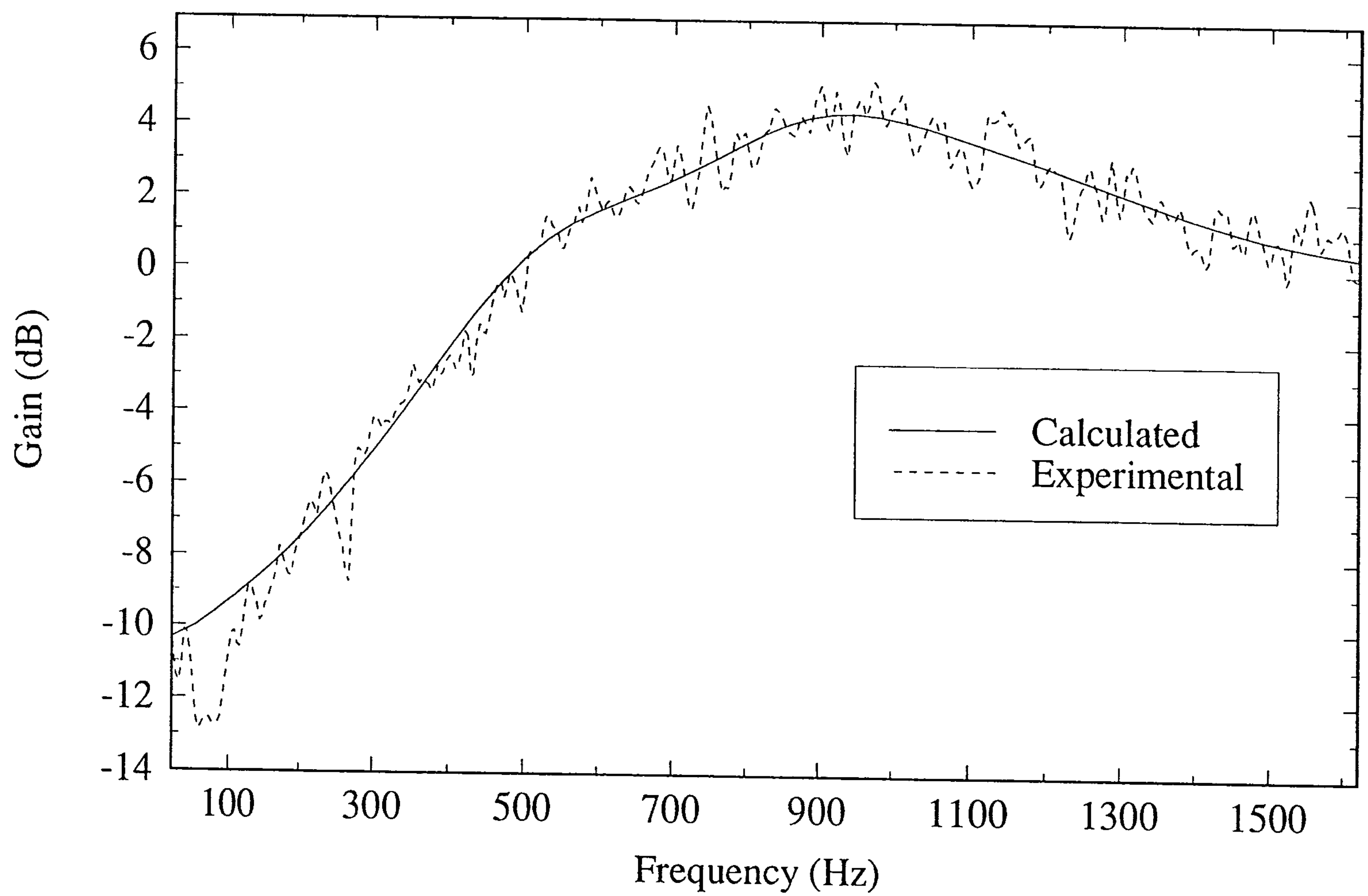


Figure 23. Comparison of Calculated and Experimental Cancellation
ITE left ear

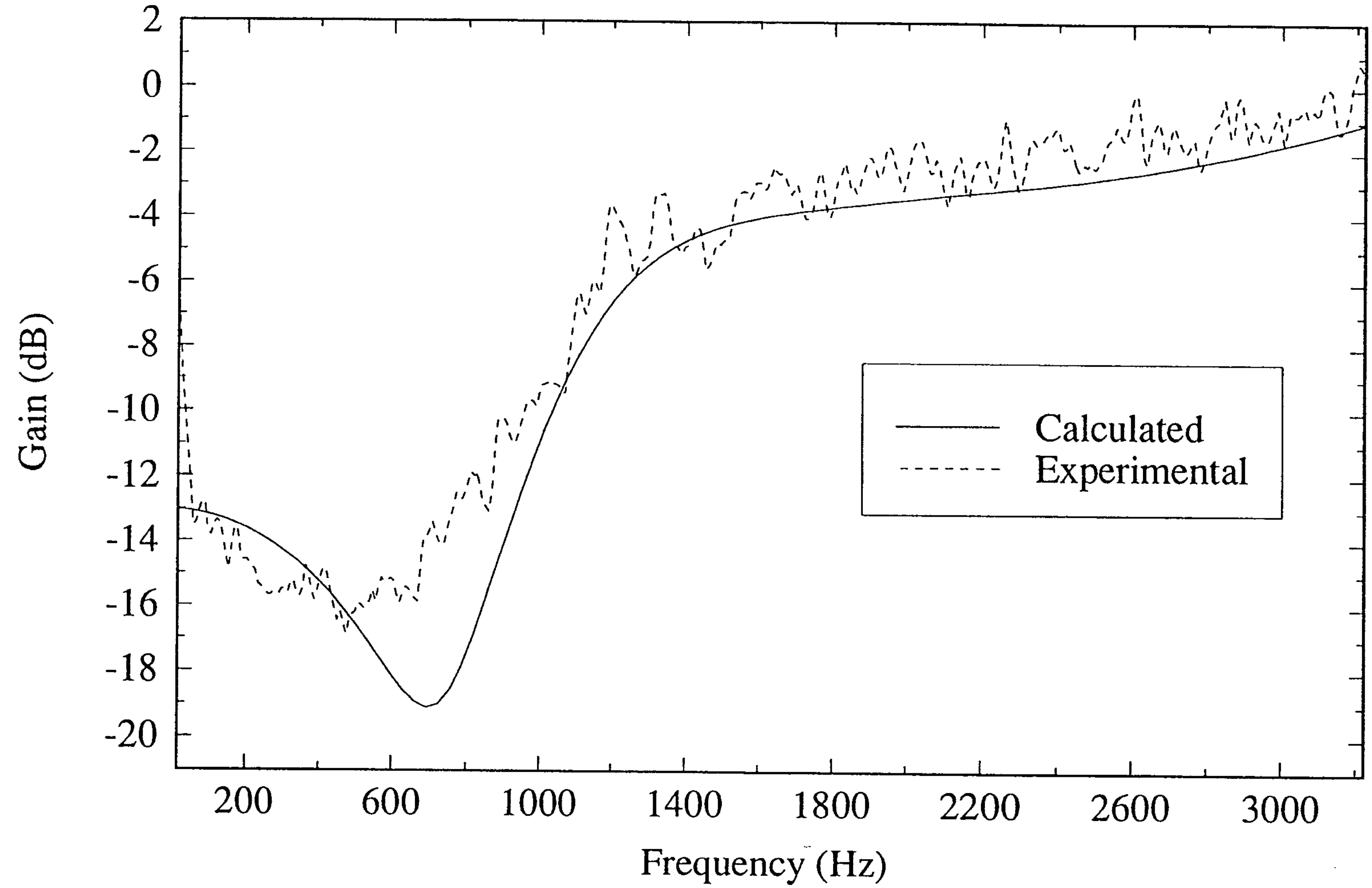


Figure 24.

	ATE	ITE
Variable Output Gain	0.381	0.381
Variable Input Gain	32.5	32.5
Programmable Gain	0.125	1.0
Ear Weighting	Flat	E
Maximal Transient Halflife	.001 sec	.001 sec
Non-recursive linear digital filter coefficients	11	15
Recursive linear digital filter coefficients	12	10
Digital filter sample period	47.9 usec	39.8 usec
Number of coefficients modelling Digital Impulse Feedback Loop Response	45	25
Numerical "effort"	3000	3000
A/D and D/A controlling DSP Internal Clock Period	6.0 usec	2.6 usec
Idle DSP time per Period	8.0 usec	12 usec
Probing Digital Impulse height	15000 (4.6 V)	32000 (9.8 V)
Number of Probes averaged	1000	2000
Number of FFTs averaged	50	50

acoustical gain indicates cancellation.

The adaptive control system can adjust to changes affecting the feedback loop, changes in the spectrum of the primary sound (to focus cancellation where it is needed), and even changes in the ear sensitivity of a specific user. The capability to adjust to changes in the feedback loop need not only be applied to compensating for gradual changes caused by a different fit, temperature variations, *etc.* One can exchange the ATE and ITE earplugs and still use the identical electronic control system. This will require, however, human guided adjustments to optimize various parameters for the chosen device. Figure 24 gives these parameters for the electronic control system. Two distinct sets of default parameters were determined at the NCPA to be appropriate for respective ATE and ITE devices. After the user designates which system is to be used, the control system automatically uses the appropriate default parameters. Variation of the first two parameters requires turning the screw of the appropriate variable resistor on the IIC circuit card. The remaining parameters are controlled by software and may be varied using a user friendly interface on a PC. It should be emphasized here that after taking the ITE data given in Fig.s 19 to 21 on 6-10-94, it was not necessary to spend another year to design and build electronic hardware for a second control system. The ITE earplug was removed, the ATE earplug was plugged in, a few parameters given in Fig. 21 were manipulated using the user friendly PC interface, and the data in Fig.s 16 to 18 was taken on 6-11-94.

The ANR control system has many other features that analog headsets do not have. These features are simple to implement in a programmable digital system. For example, if the system goes feedback unstable, it turns itself off, and will not return to operation unless the user desires to push a button to make it reset. The electronically transmitted communications signal is not affected by the acoustical gain of the negative feedback loop. Four A/D converters are in the IIC circuit card: two sample the stereo ANR error microphones and two sample the stereo analog communications signal. This digital communications signal is added to the output of the IIR digital filter. After calculating a numerical convolution of the part of the D/A output used for communications with the digital impulse feedback response coefficients, as in Eqn. 3, the part of the ANR microphone signal due to communications is known, and this quantity is digitally subtracted from the ANR microphone signal. This prevents the ANR system from trying to cancel the electronically transmitted signal.

B. Wright Patterson Air Force Base Delivery

During the week from June 13-17, 1994, the ANR control system in conjunction with the ATE and ITE ANR earplug prototypes was demonstrated at WPAFB. The control system's user friendly PC interface provides graphs of the digital impulse feedback response data, theoretically predicted Bode plots, theoretically predicted spectrums before and after ANR, ear sensitivity weighted spectrums, *etc.*

At WPAFB, for the particular jet like noise generated and particular fit of the ANR devices (affecting passive control), the primary noise for both the ATE and ITE was less band-limited than at the NCPA, making it more difficult to cancel. Starting with a spectrum resembling pink noise as closely as possible and 124.1 dB overall, the Thunder

29 ITE system had 42.2 dB of passive control. Cancellation was achieved in all third octave bands from 20 Hz to 4 kHz with an overall reduction of 3.7 dB. After shaping the spectrum to peak like jet noise at 1000 Hz the overall sound level in the reverberation room was increased to 134.1 dB. This increased ITE overall passive control to 46.1 dB, because the spectrum shaping decreased the external sound at low frequencies where the passive control is relatively weak. The ITE subsequently cancelled sound in all third octave bands from 16 Hz to 1587 Hz and an overall reduction of 6.6 dB. The 9AN/2 ATE system was put in the same jet like sound field and attained 31.9 dB of overall passive attenuation and 9.0 dB of active attenuation. For the ATE device, the primary noise was much more band limited and the cancellation band extended from 10 Hz to 400 Hz.

VIII. Conclusions

A digital negative feedback control system was developed to operate in conjunction with prototype ANR earplugs. The control system defined a digital filter to optimally reduce the ear sensitivity weighted noise at the ANR microphone subject to a relative stability constraint. The digital filter is customized to take into account the current status of the primary sound spectrum and the feedback loop.

The ANR earplugs were placed beneath passive headsets. The systems were tested on a KEMAR mannequin placed in an external sound field that simulated jet noise. This type of noise is bandlimited random noise that is dominant around 1000 Hz. In a 137 dB jet noise like external sound field, the ATE system augmented 30 dB of overall passive attenuation with 9.3 dB of overall active attenuation at the KEMAR ear drum. The ATE prototype system is fully human compatible. The ITE prototype system is not yet fully human compatible. In a 125 dB jet noise like external sound field, the ITE system augmented 40 dB of overall passive control with 11.8 dB of overall active control at the KEMAR ear drum. For higher external sound levels the ITE active control begins to degrade. At WPAFB, the ITE was tested in a 134.1 dB external SPL and 6.6 dB of overall active control was attained.

Although both the ATE and ITE systems had an optimal overall ANR "performance" of about 10 dB, the capabilities of the two systems are very different. The ATE system has relatively weak passive attenuation at low frequencies. This results in the noise under the headset and earplug being dominated by low frequencies. The additional low frequency sound makes the time delayed autocorrelation relatively large, so the the primary sound is easier to cancel. To attain the given 9.3 dB active attenuation figure, the ATE system actively reduced sound at all frequencies in the bandwidth from 10 Hz to 400 Hz. In contrast, to attain 11.8 dB overall attenuation of a much flatter primary noise spectrum, the ITE system actively reduced sound at all frequencies in the bandwidth from 16 Hz to 3175 Hz.

The results from the ATE and ITE revealed the need for future efforts in several different areas. The most important issue is that in order to improve the ANR performance, the control system should be designed to attempt to cancel noise at the eardrum rather than at the ANR microphone. As the distance between the ANR microphone and eardrum, and the frequency of sound increases, the SPL at the ANR microphone and the SPL at the eardrum can be considerably different. In particular, the ANR microphone to eardrum transfer function was different for the primary and secondary sound, so cancellation at the ANR microphone did not guarantee cancellation at the eardrum. These discrepancies undermined the usefulness of being able to chose a specific ear sensitivity weighting to be minimized. The refinement of properly weighting the sound calculated to exist at high frequencies is only useful after one is first of all able to accurately calculate the effect of ANR at the eardrum. The described digital feedback control system is particularly suitable for making this extension. In fact, no change in the hardware of the control system would be required. Digital technology will be essential for designing an ANR system that acoustically probes the user's ear, analyzes the resulting data to determine parameters affecting the ear canal acoustics, and finally determines and implements an appropriate frequency dependent modification of the feedback loop gain.

BIBLIOGRAPHY

1. G.E. Warnaka, "Active Attenuation of noise–The State of the Art", *Noise Control Engineering*, **18**[3], p.100, 1982.

2. D.C. Swanson, "A Stability Robustness Comparison of Adaptive Feedforward and Feedback Control Algorithms", *Proc. Conf. Recent Advances in Active Control of Sound and Vibration*, Technomic Publishing Co. Inc., Lancaster, PA, p.764, 1991.

3. L.E. Kinsler, A.R. Frey, et al, *Fundamentals of Acoustics* 3^{rd} edition, Wiley, New York, p.262, 1982.

4. S.S. Stevens, "Perceived Level of Noise by Mark VII and Decibels (E)", *J. Acoust. Soc. Am.* **51** p.575, 1972.

5. H.F. Olson, "Electronic Sound Absorber", *J. Acoust. Soc. Am.*, **25**[6], p.966, 1953.

6. H.F. Olson, "Electronic Control of Noise, Vibration, and Reverberation", *J. Acoust. Soc. Am.*, **28**[5], p.966, 1956.

7. H.G. Leventhall and Kh. Eghtesadi, "Active Attenuation of Noise: Monopole and Dipole Systems", *Proceedings Int. Conf. Noise Control Eng.*, S. Czarnecki, Ed., p.175, 1979.

8. S. Yamada, M. Jessel and Th. Angelini, "Active Sound Absorption by a Single Monopole Source for the Reduction of very Low Frequency Sound in a Duct", *Acoustics Letters*, **3**[2], p.56, 1979.

9. L.J. Eriksson, "Recursive Algorithms for Active Noise Control", *Proceedings from the International Symposium on Active Control of Sound and Vibration*, Acoustical Society of Japan, Osaka, Japan, pp.144–155, 1991.

10. L.J. Eriksson, *Active Sound Attenuation using Adaptive Digital Signal Processing Techniques* (thesis at Univ. of Wisconsin at Madison), University Microfilms International, Ann Arbor, MI, 1985.

11. W.K.W. Hong, Kh. Eghtesadi, H.G. Leventhall, "The tight coupled monopole (TCM) and tight-coupled tandem (TCT) attenuators: Theoretical aspects and experimental attenuation in an air duct", *J. Acoust. Soc. Am.*, **81**[2], p.376, 1987.

12. D. Andrea and J. Kowalski, "Adaptive Noise Cancellation and Speech Enhancement System and Apparatus Therefor", U.S. Patent 5,251,263, Oct. 8, 1993.

13. A.G. Bose and J. Carter, "Headphoning", U.S. Patent 4,445,675 , June 19, 1984. (see also U.S. Patents 4,494,075 4,644,581 4,856,118 5,181,252)

14. E. Langberg and F.G. Caruso, "Active Noise Reduction System", U.S. Patent 4,985,925 , Jan. 15, 1991.

15. L.J. Eriksson, M.C. Allie and R.A. Greiner, "The Selection and Application of an IIR Adaptive Filter for Use in Active Sound Attenuation", *IEEE Trans. Acous., Speech, Sig. Proc.*, **35**[4], p. 433, 1987.

16. G. Billoud, "Adaptive Active Control of Instabilities", *Proc. Conf. Recent Advances in Active Control of Sound and Vibration*, Technomic Publishing Co. Inc., Lancaster, PA, p.95, 1991.

17. L.J. Eriksson and M.C. Allie, "Use of Random Noise for On-line Transducer Modeling in an Adaptive Active Attenuation System", *J. Acoust. Soc. Am.*, **85**[2] p.797, 1989.

18. M.O. Tokhi and R.R. Leitch, "The Robust Design of Active Noise Control Systems Based on Relative Stability Measures", *J. Acoust. Soc. Am.*, **90**[1], p.334, 1991.

19. Y. Hori, "Vibration Noise Reduction Device for Electrical Apparatus", U.S. Patent 4,435,751 , March 6, 1984.

20. G.B.B. Chaplin, "Method and Apparatus for Cancelling Vibrations", U.S. Patent 4,490,841 , Dec. 25, 1984.

21. B.A. Taylor, "Method and Apparatus for Cancelling Vibrations", U.S. Patent 4,947,435 , Aug. 7, 1990.

22. C.F. Ross, *Active Control of Sound* (thesis at Queen's College, Cambridge), 1980.

23. C.F. Ross, "An Algorithm for Designing a Broadband Active Sound Control System", *Journal of Sound and Vibration*, **80**[3], p.373, 1982.

24. C.F. Ross, "An Adaptive Digital Filter for Broadband Active Sound Control", *Journal of Sound and Vibration*, **80**[3], p.381, 1982.

25. C.F. Ross, "Method and Apparatus for Active Sound Control", U.S. Patent 4,480,333, Oct. 30, 1984.

26 . TI Inc., *TMS320C4x User's Guide*, Digital Signal Processing Products of Texas Instruments, Inc., 1991.

27. B.T. Mozo, L.S. Barlow, B.A. Murphy, "Sound Attenuation Characteristics..", US Army Aeromedical Research Laboratory Report No. 93-10, January, 1993.

28. E.H. Berger, "Laboratory Attenuation of Earmuffs and Earplugs Both Singly and in Combination", *American Industrial Hygiene Association Journal*, 44[5] p.321, 1983.

29. T.R. Cuthbert Jr., *Optimization Using Personal Computers*, Wiley-Interscience Publications, New York, pp.163–178, 1987.

30 . D.M. Gay, "Algorithm 611", *ACM. Trans. on Math. Software*, **9**[4], p.503, 1983.

31. B.C. Kuo, *Automatic Control Systems* 4^{th} edition, Prentice-Hall, Englewood Cliffs, NJ, p.361, 1982.