Incorporation of loudness measures in active noise control

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An attempt has been made to use a modified version of a standard active noise control algorithm in order to take into account the unique response of the human auditory system. It has been shown in the past that decreasing the sound pressure level at a location does not guarantee a similar decrease in the perceived loudness at that location. Typically, active noise control is based on minimizing the ''error signal'' from a mechanical device such as a microphone, whose response is nominally flat across the frequency response range of the human ear. However, if the response of the ear can be approximated by digitally filtering the error signal before it reaches the adaptive controller, one can, in effect, minimize the more subjective loudness level, as opposed to the sound pressure level. The work reported here entails simulating active noise control based upon minimizing perceived loudness for a collection of input noise signals. A comparison of the loudness of the resulting error signal is made to the loudness of that resulting from standard sound pressure level minimization. It has been found that the effectiveness of this technique is largely dependent upon the nature of the input noise signal. Furthermore, this technique is judged to be worth considering for use with applications of active noise control where the uncontrolled noise more prominently constitutes low range audio frequencies (approximately 30 Hz–100 Hz) than medium range audio frequencies (approximately 300 Hz–600 Hz).
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I. INTRODUCTION

As the field of active noise control has developed, numerous applications and issues associated with active control have been investigated. However, in all of this development, there is a potentially important area that has been largely ignored. This area involves the issue of how human beings respond to the controlled field.

In implementing an active noise control system, one must of necessity use some form of sensor to obtain information regarding the acoustic field. Originally, this consisted of nothing more than a simple microphone to detect the acoustic pressure. As it became apparent that simply using a microphone could lead to undesirable results, such as localized control, other techniques began to be developed in an attempt to achieve a ''better'' solution. Such techniques include using multiple microphones to obtain a more global response,[1] ''energy density sensors'' to try to avoid local minima,[2] modal sensors to control dominant modes in the field,[3] intensity sensors to minimize propagation in a certain direction,[4] and radiation mode sensors to minimize acoustic radiation with a minimal number of structural sensors.[5] While all of these methods have been shown to have certain advantages for various applications, it has also been noted that it is often possible to implement an active control system where the performance function is attenuated substantially, but in which human observers have noted little difference between the uncontrolled and controlled states. In other words, although the mechanical sensor used in the control system detects a substantial reduction, the human ear as a sensor detects little, if any, reduction. For applications involving only stealth and/or detection, such as military applications, the response of the human ear may be of little interest. However, there are many applications where the objective of the control system is to achieve a ''better'' acoustic environment for human beings. Such examples include interior aircraft and automobile noise, active control of transformer noise, active control of highway noise, and so forth. In such applications, the only really important criterion is the human perception of the control achieved.

Recently, there has been some interest shown in addressing the question of improving an active control system in terms of human response. Saunders and Vaudrey[6] showed that a signal with significant reduction in the controlled sound pressure level may only exhibit a very modest reduction in the perceived loudness of the signal, as determined using Zwicker's method.[7] Thus their work suggested that an alternative approach to active control of sound, which could effectively reduce the loudness perceived by listeners could be useful.

This work has focused on implementing a technique that approximates the minimization of loudness by an active control system, as a means of investigating the anticipated effectiveness of such an approach. The approach taken makes use of the filtered-E algorithm, as developed by Kuo and Tsai.[8,9] Kuo and Tsai present the filtered-E algorithm as a means of altering the spectral shape of the residual noise. While they indicate the approach could be implemented based on the frequency response of the human ear, they do not investigate this possibility further. The work reported here provides a practical implementation of the filtered-E algorithm based on the response of the human ear, and evaluates how effective the method is in minimizing the perceived loudness of the

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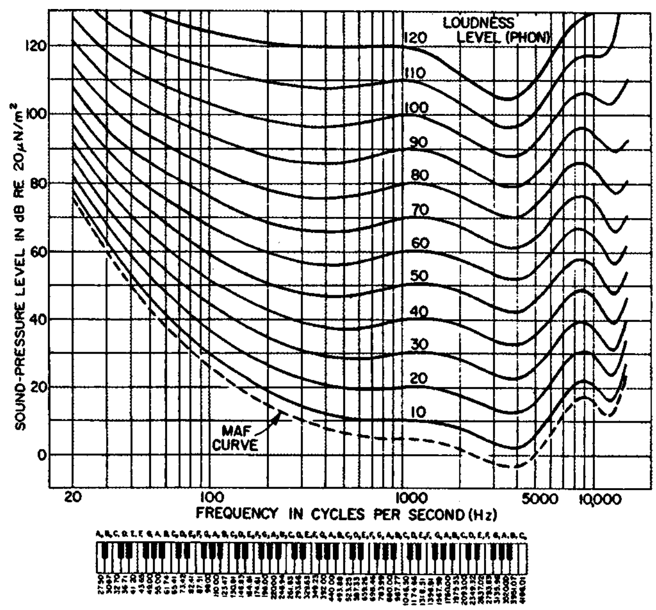

FIG. 1. Free-field equal loudness contours for pure tones, determined by Robinson and Dadson in 1956 at the National Physical Laboratory, Teddington, England (ISO/R226-1961). (Taken from Peterson, *Handbook of Noise Measurement*.)

signal. The work reported is numerical in nature, with the loudness of the controlled signals being compared for different minimization techniques as a means of judging the improvement that would be perceived by a listener.

II. DEVELOPMENT OF LOUDNESS

Loudness is a quantitative measure that is based on the subjective response of human listeners. As such, it appears that loudness could perhaps be an ideal performance function for implementing active noise control when human perception is involved. However, it is not always straightforward to calculate, and thus is difficult to implement directly in an active control system. As a result, an approximation to minimizing loudness was implemented in the active control system that was simulated for this work. However, the calculation of loudness for the various signals was still used as an analysis tool, to give an indication how well the ''loudness-based'' control could be expected to perform.

The concept of loudness was developed as a means of quantifying the manner in which the human ear responds to sound. It is well known that the human ear responds differently to the same sound pressure level presented to the listener at different frequencies. Research in this area has resulted in the development of Equal Loudness Contours, which represent the ear response over the range of human auditory sensitivity (20 Hz–20 kHz), and which can be seen in Fig. 1.

The loudness level is measured in *phons* and, like sound pressure level, is rated on a logarithmic scale. Each of the curves in Fig. 1 corresponds to a different loudness level, or phon level, chosen to be identical to the sound pressure level of the curve at 1000 Hz. Each single curve shows the sound pressure level at which various frequencies must occur in order to sound equally loud to a listener. For example, a 30-Hz sound must be at a level of about 80 dB *re*: 20 μPa to sound as loud as a 1000-Hz sound at a level of 40 dB. It is apparent that the human ear is less sensitive to low-frequency sound than to mid-frequency sound.

A distinction should be made at this point between loudness and loudness level. The loudness level is measured in phons on a logarithmic scale. Loudness is measured in *sones*, and corresponds to a linear scale. Because the scale of sones is linear, a noise signal that is measured to have one-half the number of sones as another is said to be half as loud.[10] It is possible to convert between loudness and loudness level, just as one may convert between sound pressure level (logarithmic scale) and sound pressure (linear scale).

The Stevens Mark VII procedure was used for calculating the loudness of the signals. This method utilizes all of the third-octave band levels of the signals. The corresponding perceived magnitude, S, in sones, for each band is found using a comprehensive table of values tabulated by Stevens. From the maximum, S_m, of these corresponding values a factor, F, is located in an additional table of values. The next step is to add all of the perceived magnitudes for the third-octave bands together (ΣS), and then subtract from that sum the maximum, S_m. This difference must then be multiplied by the factor F, and this product added to the maximum perceived magnitude, S_m.[11] After accomplishing these steps, the total perceived magnitude, S_t, in sones, has been determined. The preceding steps may be written mathematically as follows:

$$S_t=\left[\left(\sum S\right)-S_m\right]F+S_m. \tag{1}$$

After algebraic manipulation, this equation may be written as

$$S_t=(1-F)S_m+F\sum S, \tag{2}$$

which is the form of the equation used to calculate loudness, in sones, for this project. The value of the perceived magnitude in sones may be converted to a perceived level in decibels, if desired, by consulting an additional table developed by Stevens.

III. LOUDNESS-BASED ACTIVE CONTROL

Because there is no direct method of calculating loudness for all acoustic signals, there arises the question of how to minimize loudness with an ANC system. It is in the calculation of loudness that the Equal Loudness Contours (ELCs) become quite significant. Similar sound pressure levels at different frequencies correspond to different loudness levels (measured in phons). The difference in the loudness levels between two frequencies of similar sound pressure level is also dependent upon the sound pressure level itself; there is a trend (not without exception) such that the greater the sound pressure level, the smaller the difference in loudness levels between two frequencies of similar sound pressure level.

There already exist algorithms designed to minimize the signal received by an error microphone (namely the overall sound pressure level) in an ANC system. The filtered-x LMS

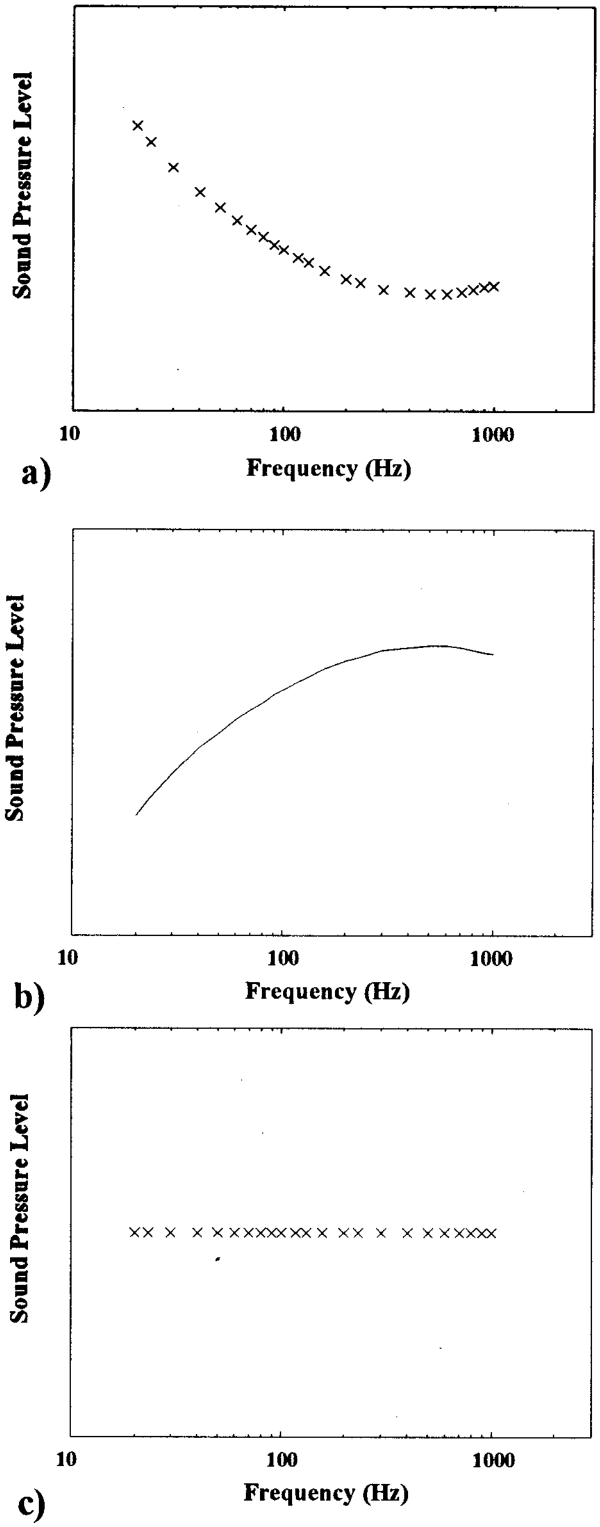

FIG. 2. Filtering operation for minimizing loudness. (a) Sample noise signal, showing relative third-octave band levels. (b) Loudness-based filter. (c) Filtered signal, normalized for loudness.

algorithm[12] is the most widely used of these algorithms. As suggested previously,[8] the error signal can be passed through a ''residual noise shaping filter,'' and then this altered signal may be used as the signal minimized by the ANC system. If this filter can be designed to have exactly the inverse shape of a given ELC, then the minimization that results will approximate the minimization of loudness associated with the

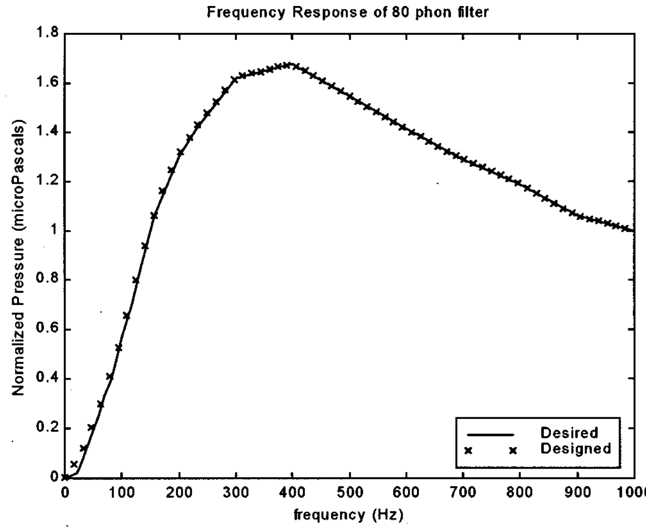

FIG. 3. 80-phon loudness-based filter. Solid line is desired filter shape. ×'s show designed filter shape.

signal. For example, since the filter is of a shape inverse to that of an ELC, then if the original signal happens to have the shape, in frequency space, of the noninverted ELC, then the filtered signal would look *flat* in frequency space. The difference between this signal and the original signal is that the levels of the frequencies in this filtered signal, which is flat in frequency space, correspond to loudness levels, as opposed to sound pressure levels. With this filtering, different frequencies are essentially normalized with respect to loudness level, and by this method, minimizing the new filtered signal by a standard method (i.e., the filtered-*x* LMS algorithm) has the effect of approaching loudness minimization of the original signal. (Note: The original signal need not have the shape in frequency space of the ELC filter used in order to attain the desired effect of filtering. This case was simply used as an example.) Figure 2 presents a simple example intended to make this concept clear. Figure 2(a) presents the third-octave band levels of a hypothetical noise signal, which has been chosen to match an ELC for the sake of illustration. Figure 2(b) indicates the loudness-based filter that can be used to properly normalize the noise spectrum. Figure 2(c) then shows the resulting filtered noise signal that could be used in an active control system designed to approximately minimize the loudness. Notice that in this example, the lower frequencies of high sound pressure level become much less significant after filtering, because the human ear is less sensitive to noise in this frequency range. For these plots sound pressure level is arbitrary.

The loudness-based filters used in the research were obtained in the following manner. The values associated with each ELC were determined, for the range of 10 phon to 120 phon, in 10-phon increments. These values, which were on a logarithmic scale, were multiplied by -1 to invert the contour, and then converted to a linear scale. The finite impulse response (FIR) loudness-based filter was then obtained using the MATLAB function ''yulewalk.'' This function uses the frequency and modulus values to obtain the FIR filter which most closely approximates the desired magnitude response. As an example, Fig. 3 shows the filter response obtained for the 80-phon loudness-based filter. A similar fit was obtained for each of the other ELCs used.

While ideally all frequencies would be attenuated to an imperceptible level, an active noise control system is always limited in performance. Suppose the signal filtered by the loudness-based filter is in fact the error signal used to update the controller parameters. The controller will in this case not utilize as many resources attenuating the high level, low-frequency content of the original noise signal as it would have, had the signal remained unfiltered.

In order to implement this technique with the LMS filtered-*x* control algorithm, the error signal, as well as the reference signal, must be filtered by a loudness-based filter, *A*, before it is used to update the controller parameters. Figure 4 shows schematically the control system in which the error signal is filtered for loudness-based control. In this block diagram, *P* represents the transfer function of the system to be controlled, *W* represents the controller transfer function, *H* represents the control path transfer function, $\mathcal{H}$ represents the model of *H*, and *A* represents the transfer function of the loudness-based filter. Notice that the filtered-*x* signal, $r(t)$, as well as the error signal, $e(t)$, is filtered by the loudness-based filter.

It is not immediately obvious that the filtered-*x* signal, $r(t)$, needs to be filtered by *A* as well, in order for the controller update scheme to work properly. However, this requirement can be illustrated mathematically in a relatively straightforward manner. It can be seen that the error signal can be expressed as

$$e(t)=d(t)+y(t). \tag{3}$$

Also, the convolution of the filter coefficients with the input signal can be represented as

$$\sum_{m=0}^{L} w_m x(t-m)=\mathbf{W}^T\mathbf{X}, \tag{4}$$

where **W** is a vector of the filter coefficients, **X** is a vector of the input data samples, and $L+1$ is the number of coefficients in the control filter. Examining the schematic in Fig. 4, it can be seen that

$$\begin{aligned} e_f(t)&=\mathbf{E}^T\mathbf{A}=\mathbf{A}^T\mathbf{E}\\ &=\sum_{i=0}^{I} a_i e(t-i)\\ &=\sum_{i=0}^{I} a_i[d(t-i)+y(t-i)]\\ &=d_f(t)+\sum_{i=0}^{I} a_i\sum_{j=0}^{J} h_j u(t-i-j)\\ &=d_f(t)+\sum_{i=0}^{I} a_i\sum_{j=0}^{J} h_j\sum_{l=0}^{L} w_l x(t-i-j-l)\\ &=d_f(t)+\sum_{i=0}^{I} a_i\sum_{l=0}^{L} w_l\sum_{j=0}^{J} h_j x(t-i-j-l). \end{aligned} \tag{5}$$

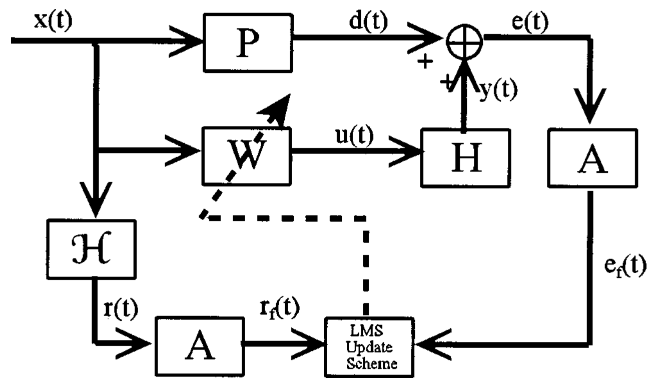

FIG. 4. Filtered-*x* scheme with loudness-based filters for $e(t)$ and $r(t)$.

Notice here that $\Sigma h_j x(t-i-j-l)$ is simply the filtered-*x* signal, $r(t-i-l)$, or in other words the noise signal, $x(t)$, after being filtered by the actuator/propagation path transfer function, *H*. Therefore, it follows that

$$\begin{aligned} e_f(t)&=d_f(t)+\sum_{i=0}^{I} a_i\sum_{l=0}^{L} w_l r(t-i-l)\\ &=d_f(t)+\sum_{l=0}^{L} w_l\sum_{i=0}^{I} a_i r(t-i-l). \end{aligned} \tag{6}$$

Because $\Sigma a_i r(t-i-l)=r_f(t-l)$, where $\mathbf{R}_f=\mathbf{R}^T\mathbf{A}=\mathbf{A}^T\mathbf{R}$, it follows that

$$\begin{aligned} e_f(t)&=d_f(t)+\sum_{l=0}^{L} w_l r_f(t-l)\\ &=d_f(t)+\mathbf{W}^T(t)\mathbf{R}_f(t). \end{aligned} \tag{7}$$

From this representation of $e_f(t)$, the new controller update equation can be formulated using standard minimization techniques as

$$\mathbf{W}(t+1)=\mathbf{W}(t)-\mu e_f(t)\mathbf{R}_f(t). \tag{8}$$

This is the update equation used in the simulations to investigate the effectiveness of using loudness-based active noise control.

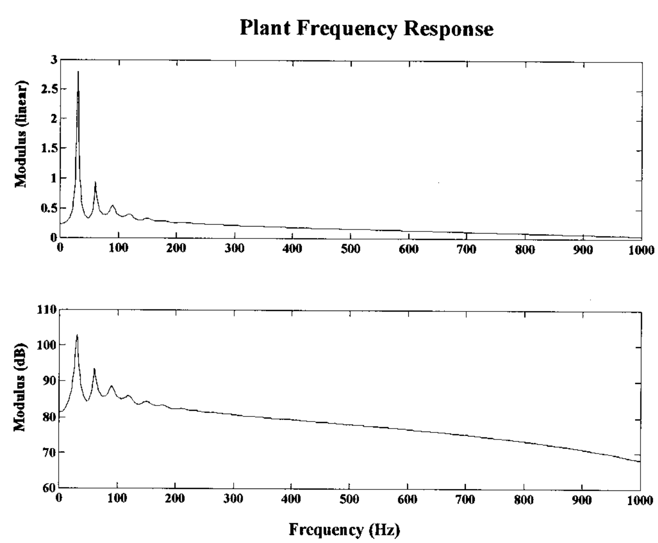

FIG. 5. Response in frequency space of the plant, *P*.

TABLE I. Best case results for SIM1, SIM2, and SIM3.

	Random noise input					
	SIM1—10 coefficients		SIM2—30 coefficients		SIM3—60 coefficients	
	L_p (dB)	Loudness (sones)	L_p (dB)	Loudness (sones)	L_p (dB)	Loudness (sones)
$d(t)$	76.9	17.0	77.4	17.1	77.1	16.9
$e(t)$	76.1	14.8	76.3	15.0	77.0	15.4
$e2(t)$	73.8	15.2	74.1	15.6	73.5	15.8

To investigate the effectiveness of the new approach, simulations were run using both loudness-based active control, and standard minimization of the squared error (pressure) signal. The plant frequency response, representing the system to be controlled, used in this work can be seen in Fig. 5. Following the simulation, the plots of the third-octave band levels superposed upon the ELCs are examined to visualize the effect of loudness control versus overall sound pressure control. Even more importantly, the actual calculated values of sound pressure level and loudness are compared for the three signals of interest, namely the two error signals based upon both methods of control, and the uncontrolled noise signal, $d(t)$. After the plots and quantitative values are examined, an assessment is made as to the apparent effectiveness of loudness control. This step may at times be somewhat difficult, due to the subjective nature of the results, although it seems clear in many cases that the difference in loudness between signals is quite significant. The simulations and assessments are repeated for a variety of input noise signals, as well as for different loudness-based filters, and for a varied number of controller coefficients. It should be noted that while the ELCs are utilized for loudness control, they do not explicitly take part in the loudness calculations. Furthermore, only the loudness calculations are used to evaluate the effectiveness of this technique.

IV. RESULTS

Throughout these comments, $d(t)$ signifies the uncontrolled noise signal as theoretically detected by the error microphone, $e(t)$ signifies the steady state error signal resulting from loudness-based control, and $e2(t)$ signifies the steady state error signal resulting from standard sound pressure level control. Furthermore, W represents the controller transfer function after loudness-based control, and $W2$ represents the controller transfer function after sound pressure level control. The transfer function used for the plant, P, nominally corresponds to a duct, and can be seen in Fig. 5. The focus of this research was on comparing minimization using the standard filtered-x algorithm with loudness-based control using the filtered-E algorithm. As a result, a simple model for the secondary path transfer function was chosen, consisting of a simple delay and gain factor $(0.99\,z^{-1})$. The convergence parameter, μ, was kept fixed at a value of 0.001 for these simulations.

For each of the simulations run, the information obtained consists of: plots of the two error signals, namely $e(t)$ and $e2(t)$, a plot of $d(t)$, FFTs of these three signals as well as of the input signal, $x(t)$, plots of the two controller transfer functions (based on the values of the controller coefficients), namely W and $W2$, and finally a single plot of the ELCs with the third-octave band levels of $d(t)$, $e(t)$, and $e2(t)$ superposed upon them. This final plot provides a visual conception of the difference between the two types of control, namely loudness-based control and overall sound pressure level control.

The first simulation that was run utilized purely random noise, generated by the ''rand'' function in MATLAB. The amplitude of random input was chosen such that $d(t)$, the

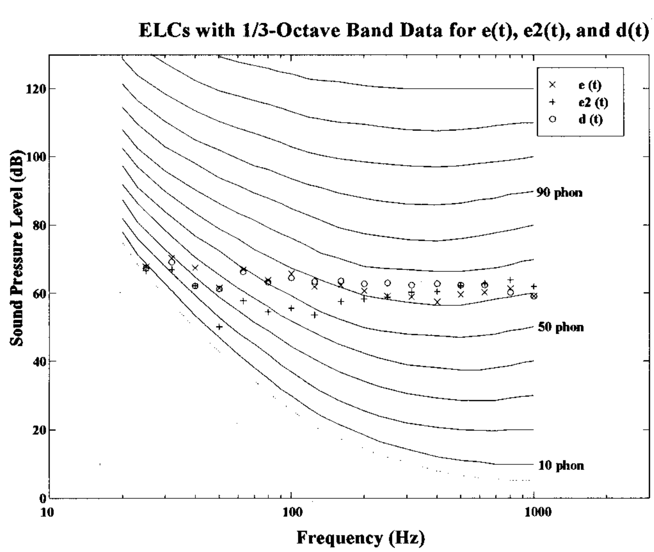

FIG. 6. Third-octave band levels of $d(t)$, $e(t)$, and $e2(t)$ for SIM1.

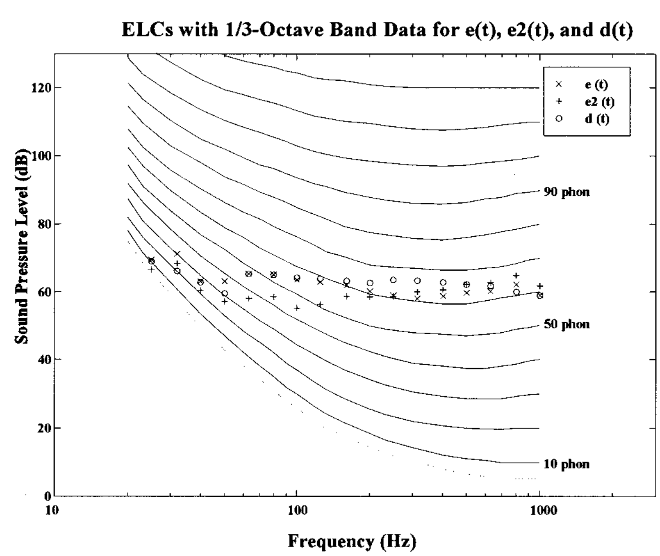

FIG. 7. Third-octave band levels of $d(t)$, $e(t)$, and $e2(t)$ for SIM2.

signal reaching the error microphone before control, constituted an overall sound pressure level of between 70 and 80 dB. Utilizing 10 controller coefficients (SIM1), this input was minimized with respect to overall sound pressure level and to loudness, using each of the 12 different phon level filters. Similar trials were run utilizing 30 controller coefficients (SIM2), and 60 controller coefficients (SIM3). The best results obtained for each of the three cases, including overall sound pressure levels and loudness for all signals, are found in Table I. These results correspond to using the 80-, 90-, and 70-phon curves respectively, which correspond closely with the level of the noise being controlled. For these results, the best case was defined as the case where the largest difference exists between the loudness of $e(t)$ and that of $e2(t)$.

Upon examination of these numerical results, little difference is found for any of the cases run with purely random input, between either the overall sound pressure level or the loudness of the two error signals. It is interesting to note, however, that in all of these cases, the overall sound pressure level of $e(t)$ is *higher* than that of $e2(t)$, while the loudness of $e(t)$ is *lower* than that of $e2(t)$. (This trend will greatly magnify in upcoming simulations, where the input signal is no longer purely random noise.) For the simulation data, the most informative plots were those showing the third-octave band levels for the uncontrolled signal, $d(t)$, as well as the levels for the loudness-based and sound pressure level control. These plots for these three random noise cases can be seen in Figs. 6–8. For this signal, it can be seen that the loudness is primarily affected by the response above 200 Hz. The loudness-based control generally provides slightly better attenuation in this frequency region, and the result is a slight improvement in the loudness of the controlled signal.

The next set of simulations (SIM4) consists simply of four sinusoids as input. It was hoped that some input signal could be created which would provide significant differences between the two methods of control, irregardless of how realistic the input signal might be. Two sinusoids were chosen at low frequencies, specifically 30 Hz and 36 Hz, where the human ear is fairly insensitive. The other two sinusoids were at frequencies of much more significant response with respect to the human auditory system, specifically 350 Hz and 400 Hz. The lower-frequency sinusoids were also chosen to be of higher amplitudes than those at the higher frequencies. This input signal was minimized utilizing 6 controller coefficients, and each of the 12 loudness-based filters. Table II presents the results for the single best case for this input signal.

TABLE II. Best case results for SIM4.

	Four sinusoids input SIM4—6 coefficients	
	L_p (dB)	Loudness (sones)
$d(t)$	71.5	2.53
$e(t)$	66.1	0.92
$e2(t)$	65.0	2.43

The results of these simulations are dramatic, especially in the difference between the loudness of the two error signals. In every case, the loudness of $e(t)$ was *significantly* lower than that of $e2(t)$. In fact, the loudness of $e(t)$ is almost always less than one-half the loudness of $e2(t)$. Furthermore, in every trial, the overall sound pressure level of $e(t)$ is *higher* than that of $e2(t)$, although not always significantly. It is interesting to note that while the overall sound pressure level was reduced in $e2(t)$ by 6.5 dB, the loudness was reduced in $e2(t)$ by a mere 4%. While one may expect to perceive a reduction of 6.5 dB, examining loudness indicates that this sound pressure level reduction would most likely be imperceptible. However, while the overall sound pressure level was reduced in $e(t)$ by 5.4 dB (for the best case), it seems very likely that, after examining the change in loudness, the difference between $d(t)$ and $e(t)$ would be very perceptible. The best case here has the loudness reduced by 62% in $e(t)$. This best case has been chosen for graphical display in Fig. 9.

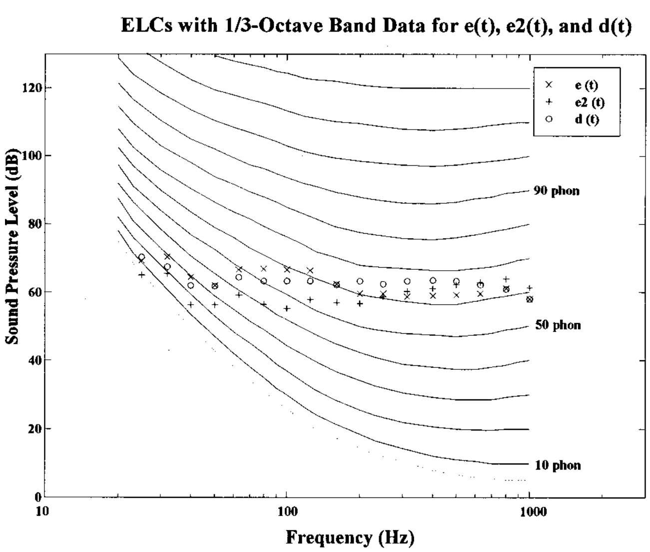

FIG. 8. Third-octave band levels of $d(t)$, $e(t)$, and $e2(t)$ for SIM3.

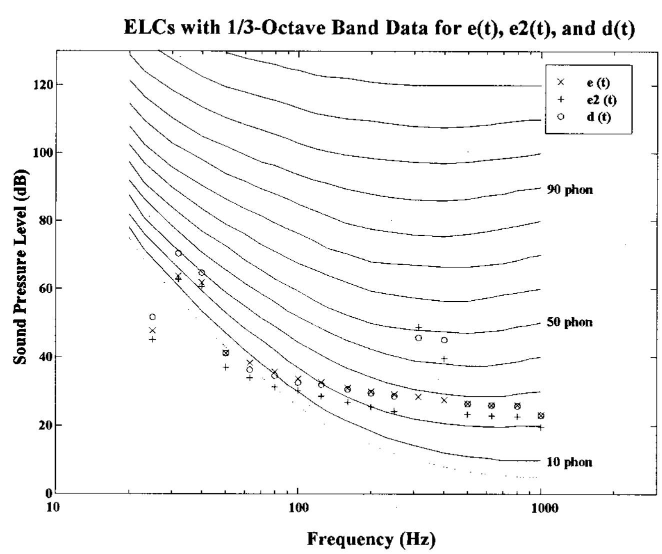

FIG. 9. Third-octave band levels of $d(t)$, $e(t)$, and $e2(t)$ for SIM4.

TABLE III. Input data for SIM7.

Frequencies included							
Freq (Hz)	Mag (Pa)	Freq (Hz)	Mag (Pa)	Freq (Hz)	Mag (Pa)	Freq (Hz)	Mag (pa)
30	0.600	210	0.255	390	0.165	570	0.0825
90	0.525	270	0.225	450	0.135	630	0.060
150	0.240	330	0.195	510	0.105	rand	±0.5

The next set of simulations combines both random and sinusoidal input. The exact frequencies and amplitudes associated with the input signal are shown in Table III. An attempt was made at generating a noise signal similar to that which might arise in a real-life situation. This signal consisted of random noise and a fundamental pure tone (30 Hz) along with odd harmonics of that fundamental tone. Each consecutive harmonic was input at a lower level than the one before (see Fig. 10), which is typical of noise signals in many different situations. Simulations were run with this input signal utilizing both 20 (SIM7) and 100 (SIM8) controller coefficients, as well as with each of the 12 different loudness-based filters. The sound pressure level of the signal $d(t)$ was between 88.5 and 89.1 dB, and the loudness was always between 15.4 and 16.3 sones. With 20 coefficients, $e2(t)$ resulted in a controlled sound pressure level of approximately 10 dB *less* than that of $d(t)$. However, the loudness of $d(t)$ *increased* in $e2(t)$ to between 18.7 and 19.8 sones, which again illustrates that while the sound pressure level has been significantly attenuated, a human observer would very likely perceive an increase in noise level. With 20 coefficients, loudness control resulted in a sound pressure level for $e(t)$ between 1 and 4 dB below that of $d(t)$, depending upon the loudness-based filter used. While these differences may seem insignificant at first, a closer look at loudness reveals some interesting results. In the best case, shown in Table IV, the loudness of $e(t)$ was lower than that of $d(t)$ by 4.2 sones. This loudness difference amounts to a decrease of almost 27%. The results for this specific result can be seen in Fig. 11.

Utilizing 100 coefficients with the same input signal finds that the loudness of $e2(t)$ is significantly lower than it was with 20 coefficients. In fact, the loudness control is slightly better in $e2(t)$ than in $e(t)$, although the difference seems insignificant. The best result here has the sound pressure level decreased in $e2(t)$ by 23 dB, and decreased in $e(t)$ by only 5.2 dB. The loudness, however, is decreased in $e2(t)$ by 6.6 sones (41%), and in $e(t)$ by 5.9 sones (37%). The difference in sound pressure level between $e2(t)$ and $e(t)$ is extreme. However, the difference in loudness is slight, further illustrating the trend that as the number of controller coefficients is increased, the difference between the two methods of control tends to decrease, specifically in regard to loudness. This last result is shown in more detail in Fig. 12.

These results are consistent with results that have been found in psychoacoustics regarding the response of the human ear.[13,14] In particular, Hellman and Zwicker[13] have shown that the loudness associated with a 1-kHz tone combined with broadband noise is not correlated with the overall sound pressure level. Some of their results indicated that one can reduce the sound pressure level by 6 dB, while simultaneously doubling the loudness. As can be seen here, similar results can be obtained when trying to minimize the loudness versus the sound pressure level.

The final set of simulations investigated were designed to examine the effect of increasing the number of coefficients made available to the controller. The trend discovered previously was such that as the number of controller coefficients increased, the difference in loudness between the two controlled signals, namely $e(t)$ and $e2(t)$, decreased. The difference in overall sound pressure level, however, tended to remain significant, and therefore the two resulting error signals were not identical, only similar in loudness. The simulations here incorporate the same input signal as in the last set (see SIM7 or SIM8), but this time the 80-phon loudness-based filter was consistently used. The number of controller coefficients was changed from 10 to 100 in increments of 10. Also included were simulations utilizing 150 and 200 coefficients. As expected, as the number of controller coefficients

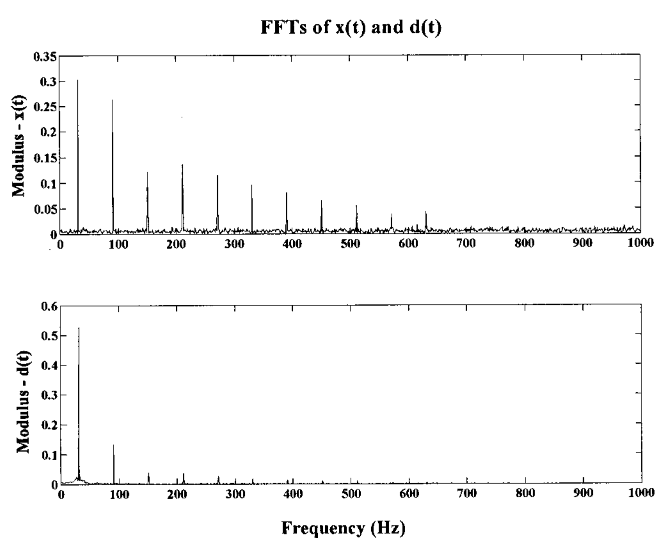

FIG. 10. FFT of $x(t)$ and $d(t)$ for SIM7.

TABLE IV. Best case results for SIM7 and SIM8.

	Many sines (30 Hz Fund.) plus random input			
	SIM7—20 coefficients		SIM8—100 coefficients	
	L_p (dB)	Loudness (sones)	L_p (dB)	Loudness (sones)
$d(t)$	88.9	15.8	88.7	16.0
$e(t)$	87.5	11.6	83.5	10.1
$e2(t)$	78.2	19.0	65.7	9.40

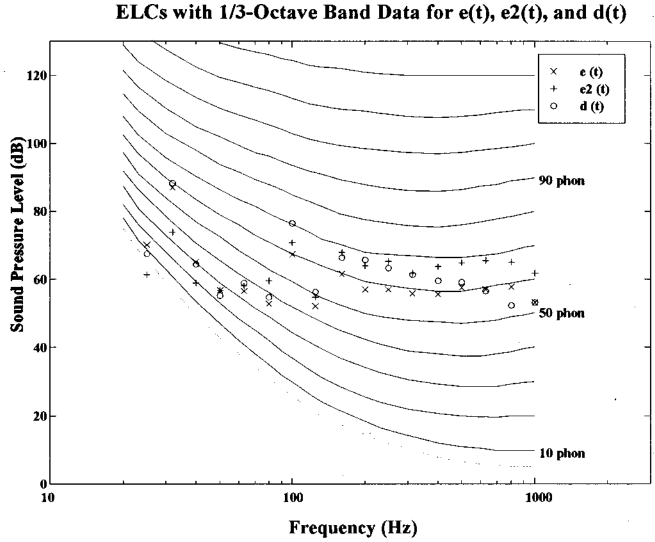

FIG. 11. Third-octave band levels of $d(t)$, $e(t)$, and $e2(t)$ for SIM7.

increased, the difference in loudness between the two error signals decreased significantly. Table V presents the results of these simulations.

The most noticeable difference occurred when increasing the number of coefficients from 30 to 40. With 30 coefficients, the difference in loudness between the two methods of control was significant. However, with 40 controller coefficients, the traditional method of sound pressure level control resulted in a loudness very similar to that resulting from loudness control. Indeed, the trend in these simulations is clear. Increasing the number of controller coefficients decreases the difference in loudness between the steady state error signals resulting from the two types of noise control. However, it is interesting to note that even when the resulting loudness is similar for the two types of control, there is a significant difference in the sound pressure level associated with the two error signals.

One possible explanation for the significant change in performance of the sound pressure level-based controller, when increasing the number of controller coefficients from 30 to 40, is that the controller has acquired enough coefficients to attempt attenuation of every sinusoid in the input noise signal. In general, the controller requires a minimum of two coefficients for each frequency it attempts to attenuate in the uncontrolled signal. These two coefficients contain the amplitude and phase information necessary for the signal matching and attenuation. Because the input noise signal for these simulations contains low-level random noise plus 11 sinusoids, the controller will not be able to attempt attenuation of all 11 sinusoids until it has at least 22 coefficients available. It appears that for this configuration, about 30 coefficients is near the minimum number of coefficients for the controller to begin to effectively attenuate all of the sinusoidal components in the signal. The loudness controller attenuates those frequencies which contribute most to the loudness of the sound, and is therefore able to achieve better loudness control with fewer coefficients.

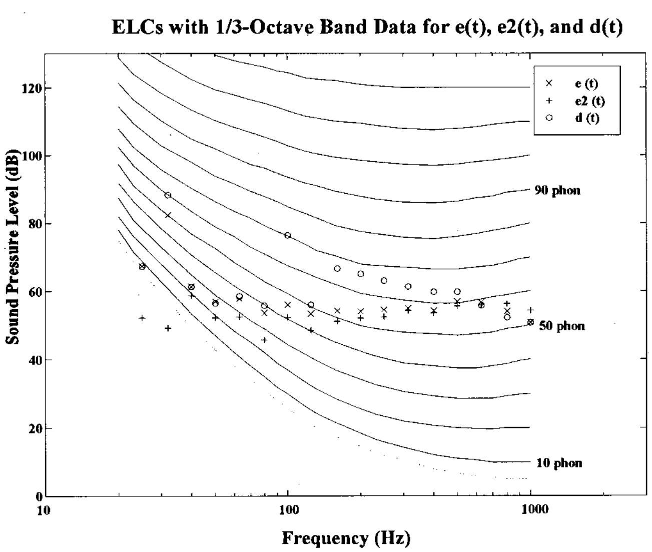

FIG. 12. Third-octave band levels of $d(t)$, $e(t)$, and $e2(t)$ for SIM8.

V. CONCLUSIONS

Since the development of the digital signal processor in the 1980's, active noise control has gained a considerable amount of attention among noise control engineers. A distinct focus of active noise control has been on applications of noise attenuation which affect the human listener. Therefore, the response of the human auditory system should at least be considered, if not explicitly incorporated, when designing the electronic controller in an ANC system.

Loudness, which is related to the human ear response, was chosen as a minimization criterion for simulations of noise control. The simulations indicate that utilizing this more subjective criterion than those traditionally used, such as sound pressure, does allow for noise control which in many cases, would likely be more pleasing to the human observer.

The apparent effectiveness of the technique employed for this project is certainly dependent upon the nature of the input signal. Several input noise signals were studied, and those that contained higher levels of low-frequency than high-frequency noise were most prominently affected by loudness control. Furthermore, because many naturally occurring acoustic signals do resemble some of the noise signals simulated during this research, it seems likely that this method would be effective in a selection of real ANC applications.

It was further noted that the number of coefficients made available to the electronic controller also affects how well this method performs. The more coefficients available, the closer the results of this technique of loudness control resemble those of traditional control, specifically in regard to loudness. Furthermore, utilizing many coefficients resulted in a significant difference in overall sound pressure level when comparing the two methods. These results are significant in that some applications of noise control have a restriction on the number of controller parameters that may be incorporated into the control system. This restriction may occur in applications involving many controllers and/or actuators, where processing time limitations exist. The results obtained here indicate that often the number of controller

TABLE V. Results which show the effect of increasing the number of controller coefficients.

Number of coefficients	Many sines (30 Hz Fund.) plus random input Sound pressure level (dB) $d(t)$	$e(t)$	$e2(t)$	Loudness (sones) $d(t)$	$e(t)$	$e2(t)$
10	89.0	88.1	80.3	15.7	12.4	19.8
20	88.8	87.8	78.4	15.9	12.2	19.1
30	88.8	88.2	74.9	16.2	11.7	18.0
40	88.9	85.7	70.9	15.8	10.1	10.8
50	89.1	85.3	68.7	16.1	10.2	10.5
60	88.8	85.7	68.7	16.1	10.2	10.5
70	88.8	84.4	66.9	15.8	9.91	9.76
80	88.9	83.6	66.3	16.0	9.88	9.63
90	88.9	84.0	65.7	15.5	9.61	9.49
100	88.9	83.4	65.4	15.4	9.92	9.25
150	88.8	80.9	64.4	15.8	9.73	9.03
200	88.9	79.2	64.2	15.9	10.44	8.98

coefficients could be substantially reduced without sacrificing the perceived loudness attenuation that is achieved.

One would not necessarily need to physically incorporate this procedure in an actual control system in order to determine the likelihood of successful implementation. By examining the frequency content of a given noise signal, as well as the constraints of the electronic controller and the physical system, one would likely be able to determine before actual experiment if a technique such as digital filtering to minimize loudness would indeed be profitable. Along with the decision to utilize this filtering technique comes the choice as to which phon level loudness-based filter should be used in conjunction with a given noise signal. Many of the results seem to indicate that there is a correlation between the third-octave band levels of the uncontrolled input noise and the phon level filter which produced the ''best case,'' although these indications are not without exception. After examination of the frequency content of the input signal, it is likely that a reasonable choice could be made as to which phon level filter would produce favorable results.

Overall, this project was found to be worthwhile, and the results practical. The real implementation of this specific procedure would not be burdensome, and may feasibly be incorporated into many ANC systems. Furthermore, because all of the results mentioned in this text are numerical in nature, true subjective effectiveness of the described loudness minimization technique may not be absolutely qualified. Research to include a panel of human observers may constitute a useful extension of this project. The subjective qualification techniques of magnitude estimation and semantic differential are two methods by which human evaluations of the effectiveness of noise control may be analyzed.[15] These methods are recommended for possible future research endeavors regarding the incorporation of subjective measures in active noise control.

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