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Hybrid Feedforward-Feedback Active Noise Control

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Abstract

This paper presents an architecture for single source, single point noise cancellation that seeks adequate gain margin and high performance for both stationary and nonstationary noise sources by combining feedforward and feedback control. Gain margins and noise reduction performance of the hybrid control architecture are validated experimentally using an earcup from a circumaural hearing protector. Results show that the hybrid system provides 5 to 30 dB active performance in the frequency range 50-800 Hz for tonal noise and 18-27 dB active performance in the same frequency range for nonstationary noise, such as aircraft or helicopter cockpit noise, improving low frequency (< 100 Hz) performance by up to 15 dB over either control component acting individually.

Introduction

Active noise reduction (ANR) has received considerable attention in the literature and in commercial products, for applications ranging from communication headsets and ventilation ducts to aircraft and vehicle cabins. The primary method employed in commercial ANR is analog feedback control, in which the signal from an error microphone serves as the feedback signal from which a cancellation signal is produced. Traditional stability-performance tradeoffs that pose limitations on feedback control of noise are evident within circumaural communication headsets, which are commercially available and serve as the focus application of this study. Passive attenuation provided by circumaural hearing protectors is minimal at low frequencies and increases with frequency. The resulting need for low frequency noise reduction motivates the use of ANR. However, in feedback ANR systems, the cavity resonant behavior forces low feedback gains and thus lower levels of active attenuation.

Figure 1 shows ANR measurements for commercial feedback headsets compiled from the literature. Included in Fig. 1 are noise reduction performance bands for feedback headsets from [1] as well as sample measurements of feedback performance from our own investigations [2]. Feedback ANR of 10-20 dB over a frequency range of 50-400 Hz is measured for stationary broadband white noise, as reported in the literature and confirmed by our measurements. However, feedback ANR often adds noise in the mid-frequency speech communication band [3,4]. This is illustrated in Fig. 1 by a *negative* noise reduction above 800 Hz.

Due to these limitations, recent research has focused on the development of feedforward ANR based on least-mean-squared (LMS) adaptive filters [5,6-10]. However, in practice, conventional LMS filters have had stability and performance deficiencies caused by (1) nonstationary, impulsive environmental noise, (2) finite precision arithmetic, and (3) measurement noise associated with quantization and electronics. The dynamic range of noise fields under which such filters must operate results in a time-varying signal-to-noise ratio (SNR) on the measured input. In response to stability issues, the family of LMS algorithms includes leaky variants [11]. The leaky LMS filter reduces ANR performance for noise sources that exhibit temporal variations over a large dynamic range because a constant leakage parameter must be selected to retain stability under worst-case SNR conditions. Moreover, tuning the leakage parameter is a highly empirical process. A Lyapunov tuning method, reported in [9,10] and validated experimentally in [9] provides a time-varying leakage factor and adaptive step size that optimizes stability of the LMS filter and noise reduction performance in response to time-varying SNR. The resulting expressions eliminate empirical tuning and add minimal computation.

Figure 1 provides sample measurements of feedforward ANR in response to individual pure tones from [6], as well as from our own work [2]. The advantage of feedforward ANR is illustrated in Fig. 1 by an increase in performance of up to 20 dB over of feedback systems in the 50-200 Hz range. Comparing Lyapunov-tuned LMS filters to feedforward system performance from [6] also shows a significant improvement below 200 Hz. The Lyapunov-tuned leaky LMS filter provides overall ANR performance for stationary noise sources that exceeds feedback performance by 10 dB on average, and it matches the performance of feedback systems to nonstationary noise sources. However, as with all feedforward systems, the ability to tolerate gain error is limited. A gain error can cause reduced performance or instability. Moreover, improved noise reduction performance for nonstationary noise is desired.

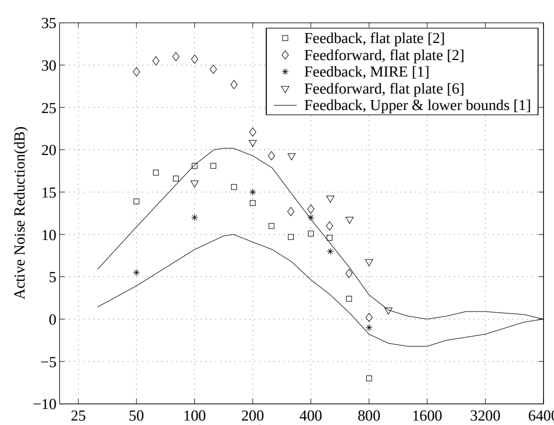

FIG. 1. ANR measurements for a commercial feedback and our prototype feedforward earcup, each with identical components, compared with published data for feedback and feedforward ANR performance.

Recent studies that consider hybrid control for acoustic noise cancellation include [12-13]. In [12], a complex filtered-X LMS filter of length 40 is combined with a commercial analog feedback controller to attenuate infrasonic noise due to the fundamental blade passage frequency in a helicopter (17.7 Hz) and its harmonics. While the feedback system, combined with passive attenuation provided by the earcup, provides 20 dB broadband performance, reduction of the fundamental blade frequency by the feedback system is minimal. Addition of the feedforward component reduces the fundamental blade passage frequency by an additional 20 dB. Reference [13] studies the effect of acoustics on ANR performance by evaluating a hybrid system in both a reverberant sound field and a directional sound field, and by exploring the relation between ANR performance and forward path delay. The results show improved performance of the hybrid system in a reverberant field, as compared with the original analog feedback system, while a directional field, which affects the acoustic delay, degrades performance when the noise source does not directly face the reference microphone.

In this paper, two additional aspects of hybrid ANR are evaluated. We present a hybrid feedforward-feedback control architecture that aims to (1) improve noise reduction performance for nonstationary noise sources, and (2) increase stability margins. First, a digital feedback system is designed to provide low level, broadband performance, independent of the noise source. The feedforward system acts on the resulting error signal to further increase noise attenuation. Unlike previous studies, where feedforward ANR is hybridized with a commercial narrowband analog controller, we develop a broadband feedback controller digitally. The presence of this feedback system is shown to increase feedforward gain stability margin substantially and to reduce sensitivity of overall performance on the temporal characteristics of the noise source. Feedback, feedforward, and hybrid performance are demonstrated experimentally using a single earcup of a circumaural headset.

Hybrid Feedforward-Feedback ANR System

Figure 2 shows a block diagram of the hybrid system. The incoming noise $X(t)$ is measured by an electret microphone embedded on the exterior of the communications earcup and digitized with an A/D converter. The past L samples of $X(t)$ constitute the reference input X_k, where L is the filter length. Electronic and quantization noise enters as Q_{xk}. As the incoming noise passes through the passive earcup, which is an unknown acoustic process, to become noise signal $d(t)$, the LMS filter finds a weight vector, $W(z)$, which is applied to X_k to produce a cancellation signal $-y_k = W^T X_k$. Meanwhile, an error microphone inside the earcup registers the current noise level, which is digitized as e_k, subject to noise Q_{ek}. The error signal serves to adjust the LMS filter, and also passes through the digital feedback compensator, $G_c(z)$, which creates its own cancellation signal $-r_k$. The two cancellation signals are scaled relative to one another by

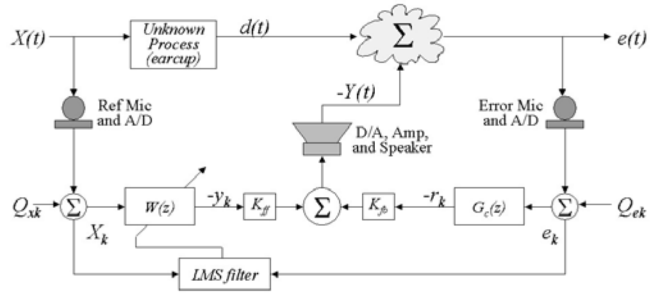

FIG. 2. Combined feedforward-feedback topology

gains K_{fb} and K_{ff}, then summed together and passed to a D/A converter. The cancellation signal is boosted by a stereo amplifier with fixed gain and is broadcast by a speaker inside the earcup as $-Y(t)$. The noise signal $d(t)$ and cancellation signal $-Y(t)$ are summed in the earcup. The residual noise, $e(t) = d(t) - Y(t)$, is what is "heard" by the error microphone, which is located about one inch from the concha, or opening to the external auditory ear canal.

Figure 3 shows the open loop transfer function from an internally generated cancellation signal $-r_k$ to the error microphone. In order to produce a broadband cancellation signal, a digital feedback compensator $G_c(z)$ was designed to increase the frequency range in which the phase response is near -180°. In this way, for a wide frequency range, the internally generated cancellation signal will be -180° out of phase with the error signal e_k. Unlike commercial headsets, which provide up to 20 dB of narrowband performance, this compensator provides lower level, broadband attenuation. In order to compensate speaker roll-off below 100 Hz, the power amp that drives the cancellation speaker (Optimus model STA-795) boosts the bass response.

References [9-10] develop the Lyapunov-tuned Least-Mean-Square feedforward algorithm. To summarize, consider the topology in Fig. 2 above. $X_k \in \boldsymbol{R}^n$ is the reference input at time step $t_k \in \boldsymbol{R}^1$, which is subject to noise Q_{xk}. The unknown process, $H(z)$, which in this case is the earcup, produces an output d_k. The LMS filter seeks a weight vector $W_k \in \boldsymbol{R}^n$ to minimize the mean-square-error the between d_k and the correction signal $W^T X_k$. The optimum or Wiener solution is $W_0 = E[X_k X_k^T]^{-1} E[X_k d_k]$, where $E[]$ denotes expected value. Following the cost function surface gradient results in a recursive weight vector update equation $W_{k+1} = W_k + \mu e_k X_k$. The convergence speed and stability of this solution is dependent on the step size μ.

In practice, the LMS filter has drawbacks. Finite precision mathematics and measurement noise can lead to instability of the weight update equation, particularly for low SNR. To improve stability, the cost function and resulting weight vector update equation are altered to leak off excess energy. Combining this leaky filter with a normalized step size μ_k gives the leaky, normalized LMS (LNLMS) filter $W_{k+1} = \lambda W_k + \mu_k e_k X_k$. , where $0 \le \lambda \le 1$ for stability. The LNLMS filter forces a stability-performance tradeoff: for high performance λ should be close to unity, whereas stability requires a larger amount of leakage (lower λ). The optimum λ changes with signal strength and

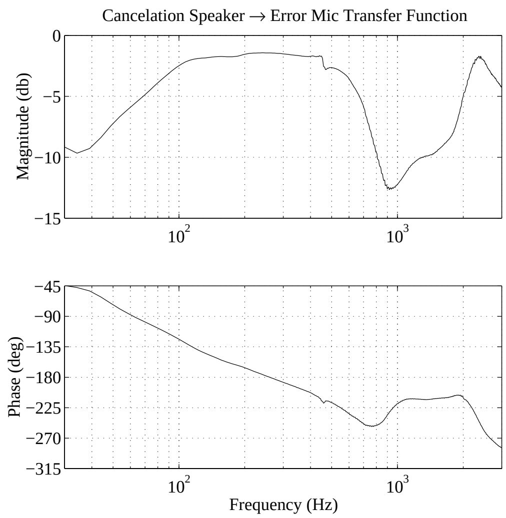

FIG. 3. Open loop transfer function from an internally generated signal, through the cancellation speaker, to the error microphone

spectral composition. A solution to this tradeoff is to impose an adaptive leakage factor. The Lyapunov tuning method [9,10] provides an adaptive leakage factor and step size combination to optimize both stability of the weight update equation and noise reduction performance, with

$$\lambda_k = 1 - \frac{2L\sigma^2}{\|X_k + Q_k\|} \quad \mu_k = \frac{\mu_o \lambda_k}{\|X_k + Q_k\|} \qquad (1,2)$$

σ^2 is the variance of the measurement noise in the reference microphone path, which can be determined empirically, and μ_o is a step size parameter. Table 1 demonstrates how this feedforward algorithm allows the leakage factor to adapt to changing noise levels and sources. As shown in [9-10], this solution enhances stability and performance over traditional LMS filters, even for nonstationary noise.

Experimental Configuration and Procedure

Experimental evaluation of the hybrid feedforward-feedback system was conducted in a Low Frequency Acoustic Test Cell (LFATC), described in [14]. The test cell acts as a one-dimensional waveguide and is designed to have a flat (to ±1 dB) acoustic frequency response from 10 to 200 Hz. Digital equalization extends this range to approximately 1600 Hz. A single earcup is mounted over the base plate of the test cell with an airtight seal. The test cell instrumentation includes: (1) a 15.2 cm diameter 100 W speaker mounted in the top plate of the cell to provide the noise signal (up to 140 dB); and (2) two precision Brüel & Kjær 4190 Type I microphones. One microphone is mounted through the sidewall of the test cell for source level measurement and the other is mounted axially in the base plate under the earcup to represent the location of the external opening to the ear canal. Noise floors of these precision microphones average 53 dB and 48 dB, respectively, in the measurement range 40-1250 Hz. The external precision microphone has a higher noise floor due to hiss from the 100 W source speaker.

The test device consists of an earcup taken from a commercial feedback ANR headset. Existing hardware within the earcup includes a noise cancellation speaker, an electret error microphone, a communication speaker (not used in this study), and feedback ANR circuitry (also not used in this study). Without disturbing the damping materials that provide passive noise attenuation, a 0.500" hole was drilled in the shell to add an external reference electret microphone. The two microphones are conditioned through preamplifiers developed in-house, which provide a noise floor of 50 dB in the measurement range 40-1250 Hz and a dynamic range of at least 75 dB. When mounted on the base of the test cell, the earcup's error and reference microphones are calibrated with respect to the precision Brüel & Kjær microphones mounted in the base and side of the test cell.

Four noise sources were selected for the performance evaluation: (1) Individual pure tones at 1/3-octave center frequencies from 40 Hz through 1250 Hz (40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, and 1250 Hz), (2) a sum-of-tones signal comprised of 1/3-octave pure tones between 50 Hz and 800 Hz, (3) F-16 cockpit noise band-limited between 50 Hz and 800 Hz, and (4) Huey helicopter noise likewise band-limited between 50 Hz and 800 Hz. These noise sources can be viewed as increasingly less ideal operating conditions. Pure tones are the most ideal operating condition as they allow the cancellation gains K_{ff} and K_{fb} to be optimally tuned for each frequency, whereas for all other noise sources one value for each gain is applied to all frequencies. F-16 aircraft noise is similar to band-limited pink noise in that it has components from all frequencies in the range 50-800 Hz. However, the two minute noise source recording used in experiments exhibits significant temporal variation. Huey helicopter noise also resembles pink noise, but with the addition of both tonal components (a 55 Hz fundamental and associated harmonics) and impulsive staccato components in the time domain from the rotor blades passing 10.7 times per second. The first three noise sources are set to an average level of

	Mean Leakage Factor λ			Std Dev (λ) x 1,000,000		
Noise Source	80 dB	100 dB	110 dB	80 dB	100 dB	110 dB
Sum-of-tones 50-800 Hz	0.976 549 014	0.999 741 113	0.999 974 634	2517.412	27.791	2.604
F-16 Cockpit 50-800 Hz	0.976 185 441	0.999 701 619	0.999 969 884	4720.885	70.043	8.008
Huey Cockpit 50-800 Hz	0.969 621 673	0.999 632 597	0.999 969 232	21273.195	263.522	19.110

Table 1. Mean value of λ_k for three noise sources, showing adaptation of LMS filter to changing SNR conditions

110 dB, whereas the fourth noise source (Huey helicopter) is set to 105 dB to avoid distortion in the cancellation speaker. All noise levels are reported in dB relative to a reference pressure of 20 μPa (the limit of human hearing), with no weighting applied.

Using this experimental setup, passive, active and total ANR performances are measured using the precision microphones. The hybrid controller is implemented using Simulink and is compiled to run in real-time on a dSPACE DS1103 controller board residing within a host computer. The controller board is based on a PowerPC 604e microprocessor running at 400 MHz. The dSPACE board handles input and output with on-board 16-bit A/D and 12-bit D/A converters. The hybrid system operates at an update frequency of 10 kHz; the LMS filter length is 500 taps. Data recorded from the four microphones are first bandpass filtered between 40-1250 Hz. Antialiasing filters for input and output channels have not yet been implemented, thus all results are presented within this band. Unless otherwise noted, all reduction performance data are given as the insertion loss between the precision microphone outside of the earcup and the one inside in the base of the test cell. Thus, they account for the separation path between the noise source and the wearer's ear.

Experimental Results

Figure 4 shows the active attenuation in dB for individual tones, as measured by the B&K precision microphone located inside the earcup. *Active Attenuation*, as defined throughout these results, is the difference between the noise level with passive attenuation and the noise level with both passive and active attenuation. It is the amount of attenuation, added to the passive system, that the active system provides. The results show that the feedback system has low level (5-10 dB) but high bandwidth noise reduction capabilities. In contrast, the feedforward system performs exceptionally well in the range 80-400 Hz, with diminished performance above and below that range. Whereas both systems have only moderate to good attenuation at low (< 100 Hz) frequencies, the combined system is able to provide approximately 30 dB of active attenuation at these frequencies. *Combining the two independent systems has resulted in performance that is greater than the sum of its parts*. Additionally, whereas both the feedforward and feedback systems add noise above 700 Hz, the combined system provides positive attenuation throughout the frequency range 40-1250 Hz. The result is that while the source volume lies at 110 dB outside the earcup, the noise level inside has been reduced by a total of 36 to 51 dB within the 40-1250 Hz band. Total noise reduction, which include passive attenuation, causes the error microphone signal to approach its noise floor, thus the noise reduction performance approaches its physical limits.

Figure 5 shows the active attenuation of the sum-of-tones noise source. In this and subsequent cases, the feedforward and feedback gains can assume only one value for all frequencies, whereas for individual tones, K_{ff} and K_{fb} can be tuned to an optimal value for each frequency. The results show that the individual tones are successfully attenuated by as much as 28 dB by the hybrid system. The hybrid system exhibits the same synergistic performance improvement over the independent feedforward and feedback systems. Whereas the feedback system provides an average of only 7.8 dB of active attenuation, and the feedforward an average of 16.6 dB, the hybrid system provides an average of 27.2 dB of active attenuation. When combined with the earcup's passive attenuation, this means that an average source level of 110 dB is successfully reduced to 70.6 dB, a level that is considered safe for long periods of exposure.

Figure 6 presents the active attenuation for each system when subjected to F-16 aircraft noise. This noise source most closely resembles band-limited white noise in that it contains no purely tonal content and, over long time periods, has a fairly uniform spectral component. However, during short periods of time its spectral content shifts considerably, which presents problems for traditional LMS filter designs. Despite the difficulties associated with this nonstationary noise source, the results show that the hybrid system provides an average active attenuation of 17.3 dB (32 dB total attenuation), reducing the 110 dB source level to 78.2 dB. Once again, the results show that the hybrid system has substantially greater performance than either of the independent systems acting alone, particularly for frequencies less than 200 Hz. Additionally, whereas the feedforward system *added* noise for frequencies above 500 Hz, the hybrid system largely avoided adding any noise in the 50-800 Hz band.

Lastly, the systems were subjected to Huey helicopter noise. This noise source contains broadband nonstationary components like the F-16 noise, but also has a tonal component following a 55 Hz fundamental attributed to the tail rotor, and a temporal component - the *thwt-thwt-thwt* of the blade passage. This temporal component is more like a periodic broadband impulse, rather than a low-frequency harmonic (as it is in [12]). In order to keep this periodic impulse from forcing the ANR systems to over-drive the cancellation speaker, the source level is reduced to 105 dB. The active attenuation results are shown in Figure 7. Once again, the addition of the feedback system to the feedforward system significantly improved the low-frequency attenuation, in this case by 5-10 dB. The tonal component is eliminated by both the feedforward and hybrid system, but largely untouched by the feedback system. The feedback system was unsuccessful in removing the temporal component of the helicopter noise; the feedforward system could not completely remove it, either. In contrast, the combined hybrid system is able to almost completely remove the periodic *thwt*, leaving behind a broadband background noise whose average level was 77.4 dB.

Table 2 summarizes these performance results, showing average source, passive, active, and total noise reduction performance for each noise source.

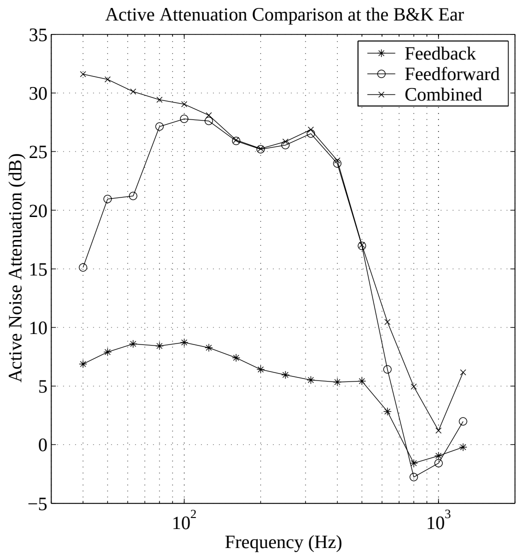

FIG. 4. Active Attenuation performance of each ANR system in response to puretone noise.

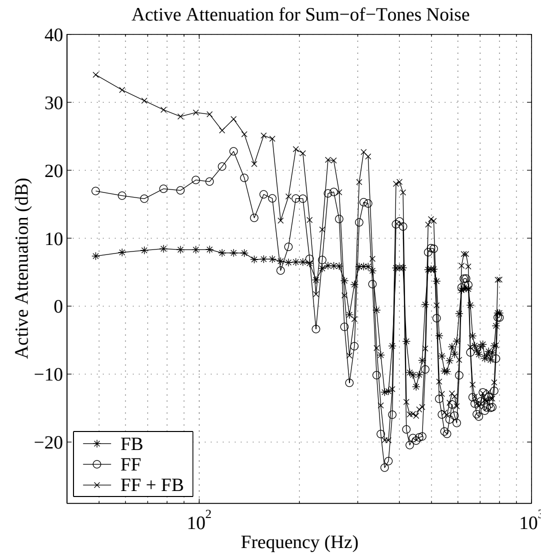

FIG. 5. Active Attenuation performance of the ANR systems when subjected to Sum-of-Tones noise

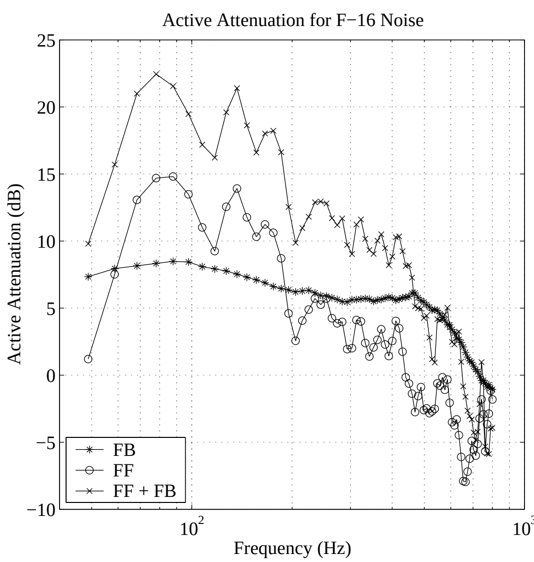

FIG. 6. Active Attenuation performance of the ANR systems due to F-16 cockpit noise

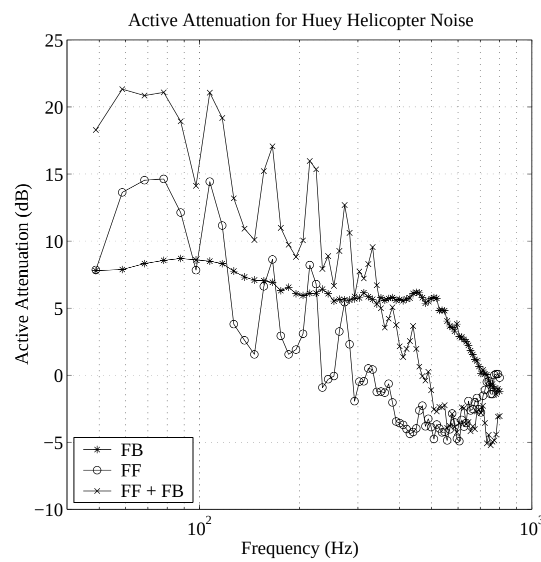

FIG. 7. Active Attenuation performance of each ANR system due to Huey Helicopter noise

The addition of feedback to the feedforward system in the hybrid system not only improves active performance, but it also improves the gain margin of the individual systems. In the feedback system, increasing the path gain, K_{fb}, generally increases the feedback attenuation. However, the gain that provides maximum noise attenuation is extremely close to the threshold of instability, forcing a stability-performance tradeoff common in commercial feedback systems. In a similar way there is a maximum feedforward gain, K_{ff}, above which the weight vector *W(z)* grows without bound, or overexcites certain frequencies (particularly 700-800 Hz). However, when the two systems are combined, both gains can be increased to levels that otherwise would cause instability. When this happens, the increased gain allows for higher overall active attenuation. Stated differently, the feedforward and feedback systems in the hybrid system can provide the same attenuation levels at the same gain values as before, but now have larger stability margins. For the feedback system, adding the feedforward system allows K_{fb} to be increased by approximately 20% before instability reoccurs. However, the increased stability is most notable in the feedforward system. Figure 8 shows the *maximum stable* (not necessarily optimal) feedforward gain K_{ff}, as determined experimentally, as a function of frequency. As Fig. 8 shows, augmenting the feedforward system with feedback allows the maximum stable K_{ff} to be increased by, at some frequencies, orders of magnitude. The fact that the maximum stable gain rolls off below 200 Hz is due to bass boost in the stereo amplifier that drives the cancellation speaker.

Noise Source	Average Noise Level (dB)					Total Attenuation (dB)			Active Attenuation (dB)		
	Source	Passive	Feedb.	Feedf.	Hybrid	Feedb.	Feedf.	Hybrid	Feedb.	Feedf.	Hybrid
Sum-of-tones 50 - 800 Hz	110.3	97.8	90.0	81.2	70.6	20.3	29.1	39.7	7.8	16.6	27.2
F-16 Cockpit 50 - 800 Hz	110.3	95.4	87.6	85.6	78.2	22.7	24.6	32.1	7.9	9.8	17.3
Huey Cockpit 50 - 800 Hz	105.3	94.2	86.0	83.7	75.8	19.3	21.6	29.5	8.2	10.5	18.4

Table 2. Summary of passive, active, and total noise reduction performance for feedback, feedforward, and hybrid ANR

As prior studies show, a frequency-dependent feedforward gain, provided by a filtered-*X* LMS system, rather than a scalar constant, would improve performance [12,13]. Future investigations will focus on a filtered-*X* implementation of K_{ff} in combination with the Lyapunov-tuned leaky LMS filter.

Conclusion

A hybrid active noise reduction system was developed that combines a Lyapunov-tuned LMS feedforward filter with a digital feedback system. The results show that the hybrid system has considerably better noise reduction performance and stability than either individual system. At low frequencies, the hybrid system's noise reduction performance is greater than the sum of that of each independent system. When subjected to a variety of noise sources, both stationary and nonstationary, the combined hybrid system provides between 30 and 40 dB of total attenuation. Future investigation will focus on developing a binaural ANR system, filtered-*X* implementation of the feedforward cancellation path gain and subject testing in both reverberant and directed sound fields.

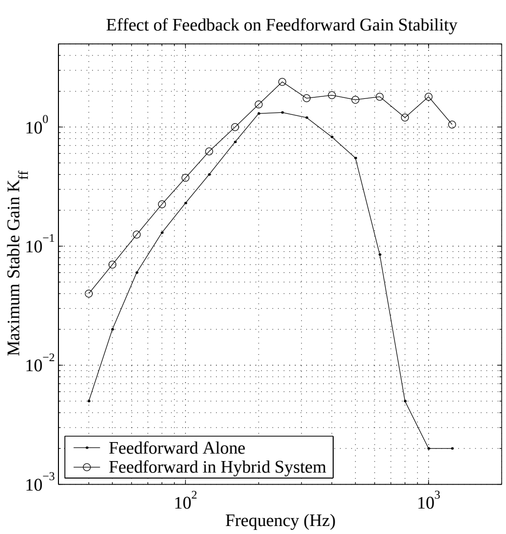

FIG. 8. Maximum stable gains (K_{ff}) of the feedforward system

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