NASA-Ames Award No. NAG2-1544 "Development of a Laboratory for Improving Communication between Air Traffic Controllers and Pilots"

SUMMARY OF RESEARCH

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Negative Inventions Statement: No inventions resulted from the work performed under this award.

Introduction

Runway incursions[1] and other surface incidents[2] are known to be significant threats to aviation safety and efficiency. Though the number of near mid-air collisions in U.S. air space has remained unchanged during the last five years, the number of runway incursions has increased and they are almost all due to human error. The three most common factors contributing to air traffic controller and pilot error in airport operations include two that involve failed auditory communication. This project addressed the problems of auditory communication in air traffic control from an acoustical standpoint, by establishing an acoustics laboratory designed for this purpose and initiating research into selected topics that show promise for improving voice communications between air traffic controllers and pilots.

The Acoustics Laboratory

The acoustics laboratory consists of an anechoic chamber, a reverberation room and two supporting laboratories. A small control room separates the anechoic chamber from the reverberation room to house electronic equipment associated with the sound sources for the reverberation room.

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[1] An occurrence involving an aircraft, other vehicle, person or object that creates a collision hazard or loss of separation with an aircraft taking off, intending to take off, landing or intending to land.

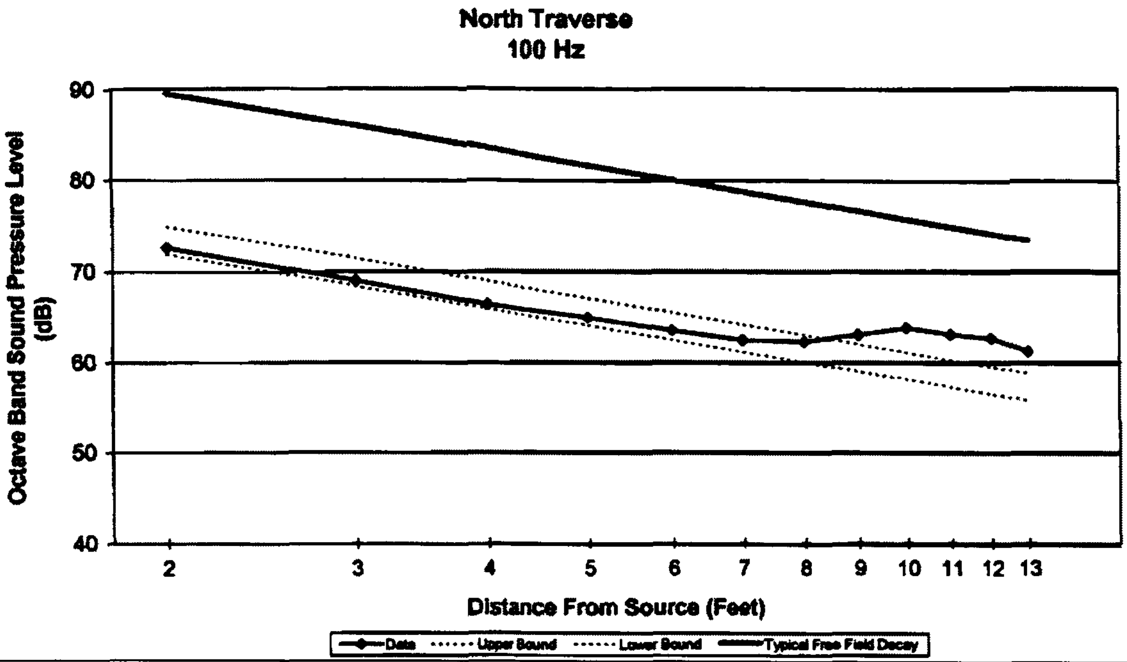
[2] Any event where unauthorized or unapproved movements occur within the movement area or an occurrence in the movement area associated with the operation of an aircraft that affects or could affect the safety of flight.

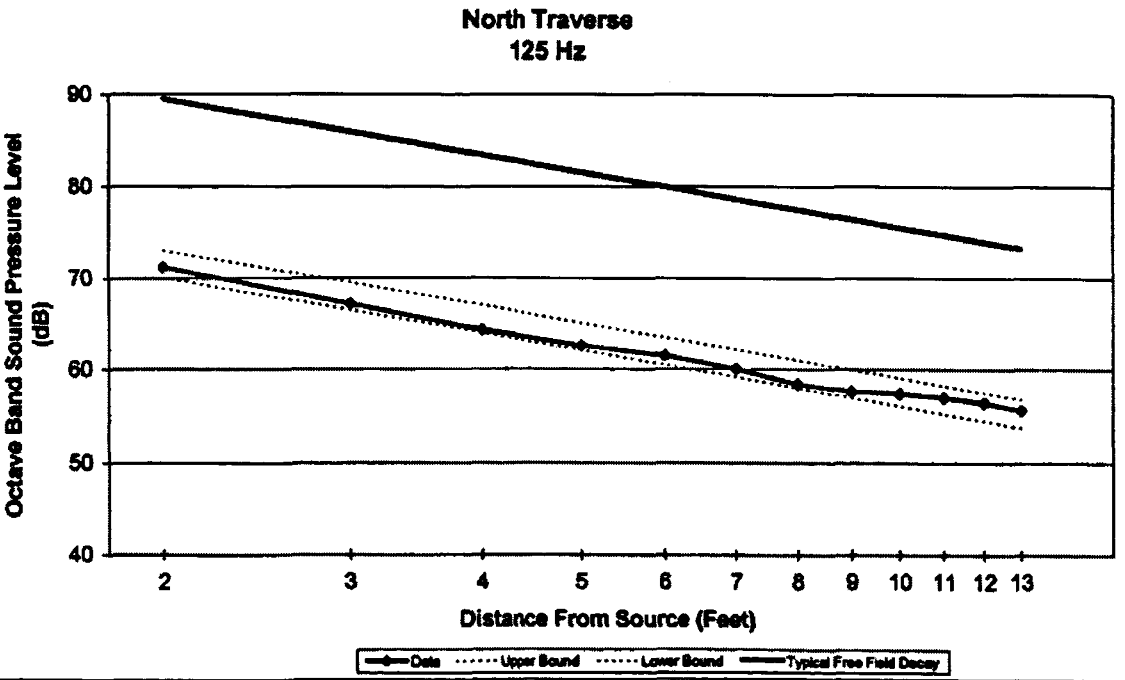
All surfaces of the anechoic chamber are lined with fiberglass wedges designed to be anechoic up to a frequency of 200 kHz, with a low-frequency cut-off frequency of 110 Hz. There is a removable grating floor (see photograph, Fig. 1). The space between the wedge tips is approximately 30'x14'x13'.

Figure 1: View of interior of anechoic chamber

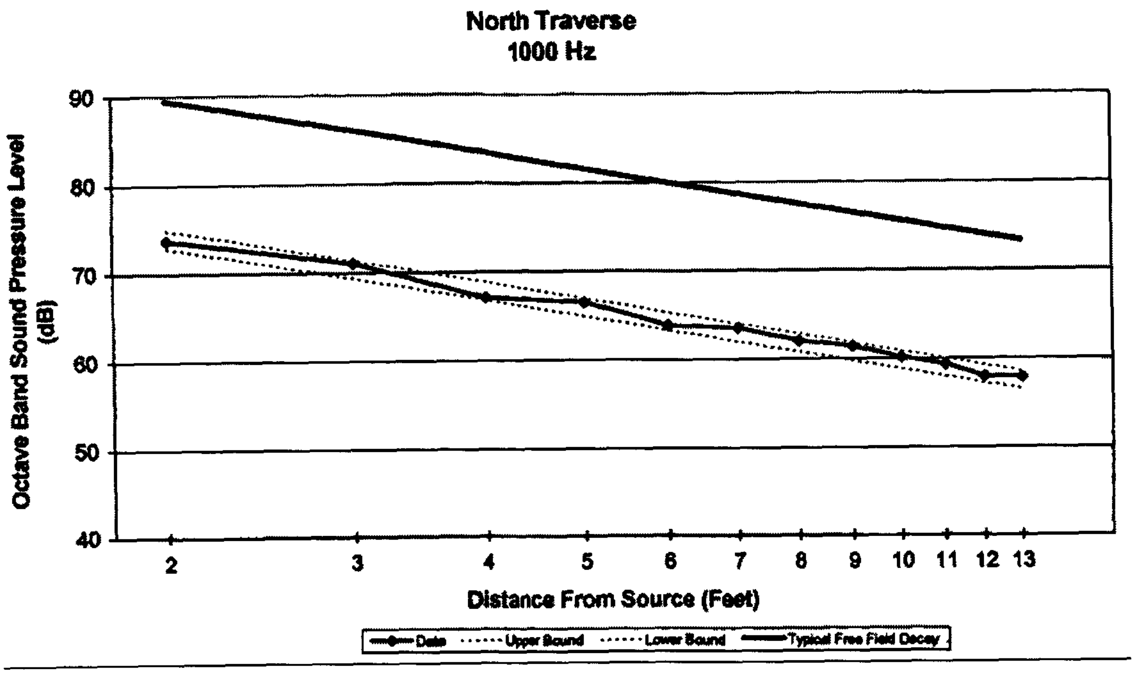
The chamber was designed to comply with the performance requirements of ISO 3745 (1977). The sound pressure decrease with distance from a compact loudspeaker driven with pink noise, or an omni-directional compressed air driven nozzle (a high-frequency broadband sound source), was determined for several directions of sound propagation relative to the wedge tips. The source was located approximately in the middle of the chamber. Typical results are shown for low frequencies in Figs. 2 and 3, and for mid and high frequencies in Figs. 4 and 5, respectively. In these diagrams the measurements were taken along the length of the chamber, parallel to the wedge tips, and so will detect reflections from the end walls (a common problem at low frequencies in anechoic chambers). The expected free-field decay of the sound field within the chamber is shown in Figs. 2-5 by thick continuous lines, for an arbitrary source strength. The sound pressure levels (the ordinate) observed as a function of distance from the sound source (the abscissa) are shown by the diamonds and the thin continuous line. Deviations from free-field performance allowed by ISO 3745 are indicated by the dashed lines.

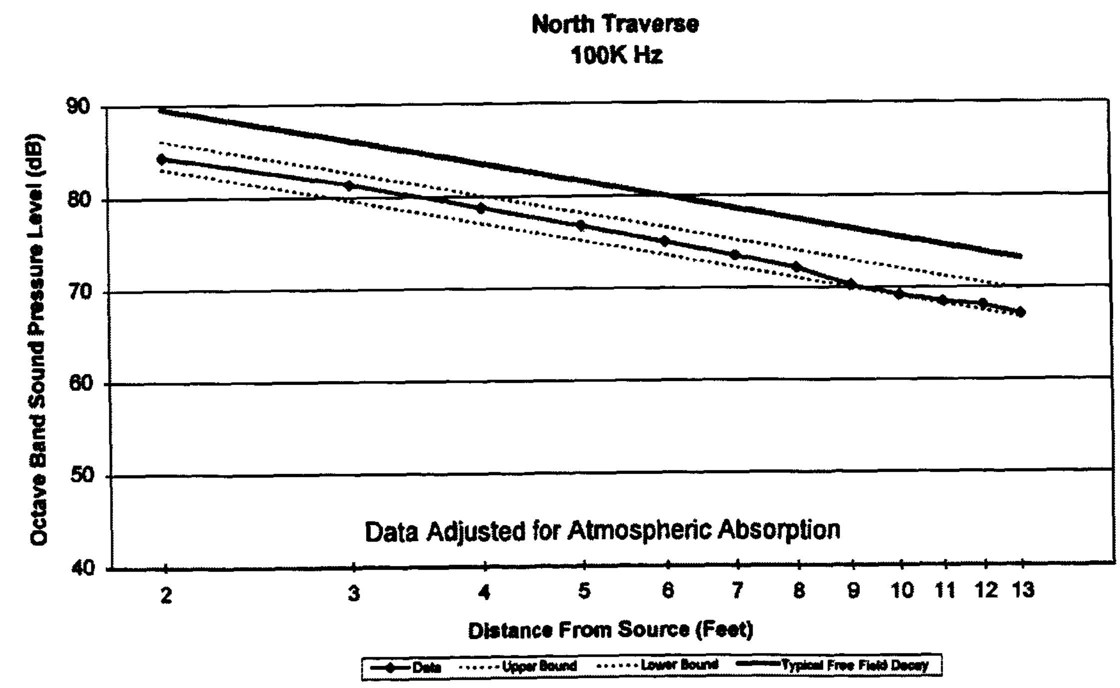
Figures 2 and 3: Low-frequency free-field performance of anechoic chamber





Figures 4 and 5: Mid- and high-frequency free-field performance of anechoic chamber

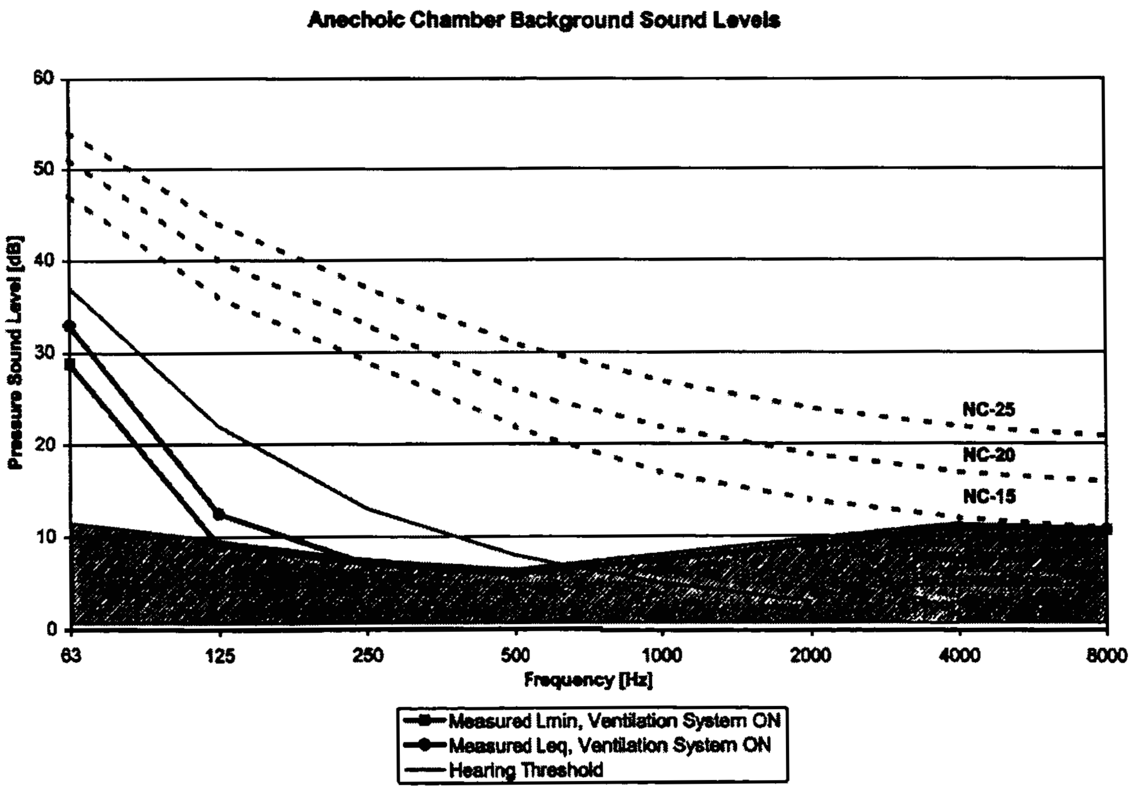




It is evident from the data in Figs. 2-5 that the sound field decay remains within the allowed range (i.e., between the dashed lines) at frequencies of 125 Hz and above, after adjusting the data at 100 kHz for atmospheric sound absorption along the propagation path, as expected from the design specifications. The sound pressure decrease with distance deviates from free-field performance at 100 Hz (at a distance of approximately 8' from the source, see Fig. 2), which is acceptable as this frequency is below the design cut-off frequency for the anechoic chamber.

The residual background noise in the chamber is required to be below the threshold of hearing when the heating, ventilation and air conditioning (HVAC) systems are operating. The measured performance is shown in Fig. 6. The (electronic) instrumentation noise imposes a "floor" on the measurements, and is shown by the shaded area. The observed sound pressures are reported for octave bands with center frequencies from 63 Hz to 8 kHz, and have not been corrected for instrumentation noise, They are shown by the filled circles and squares, and the thick continuous lines, and lie below the contour for the threshold of hearing (thin continuous line) at frequencies for which meaningful measurements can be made (i.e., below 125 Hz).

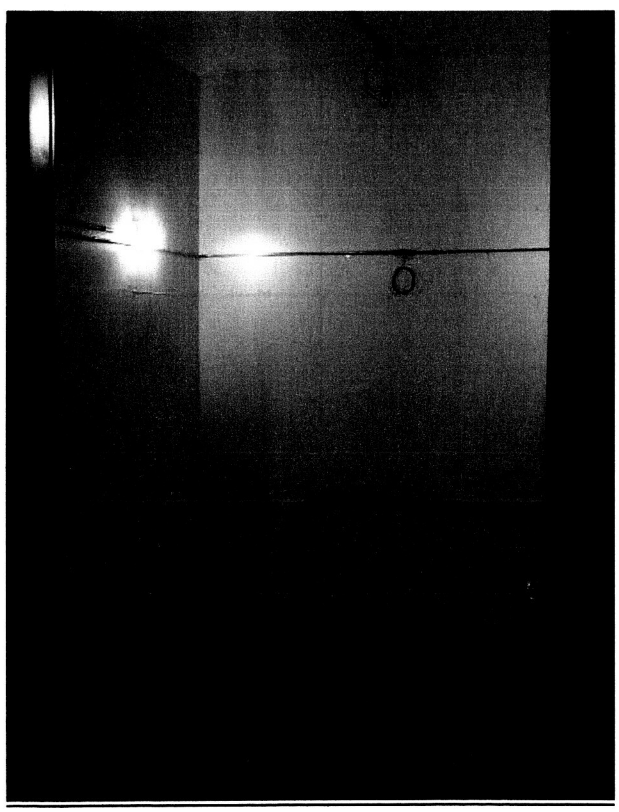
Figure 6: Background sound pressure level in the anechoic chamber



The measurements summarized in Fig. 6 were conducted during last summer. During the winter months some listeners have reported that they can hear a tonal noise in the anechoic chamber at a frequency of about 250 Hz. This is not evident in the data of Fig. 6. Further measurements will need to be conducted using instrumentation with lower equipment noise to establish whether or not the HVAC system noise changes with the seasons.

A companion reverberation room with an estimated diffuse field low frequency cut-off of 110 Hz is adjacent to the anechoic chamber. The reverberation room is designed in accordance with the requirements of ISO 3741 (1975), and has a sealed concrete floor, with walls and ceiling of melamine-skinned particleboard glued and screwed to gypsum board, and supported by metal studs. The rectangular room has dimensions 20'x13'x16.5', and is entered at about 4' above floor level (see photograph, Fig. 7). The 16.5' ceiling ensures sufficient volume for the reverberation room and provides, in the upper half, a large volume for mounting sound sources positioned and oriented to ensure measurements (in the lower half of the room) are outside the direct sound field.

Figure 7: View of reverberation room through entry door



A machine shop and an electronic shop are located in space made available in the basement of another building. All facilities are equipped with the electronic instruments and machines specified in the original proposal.

Topics for Research

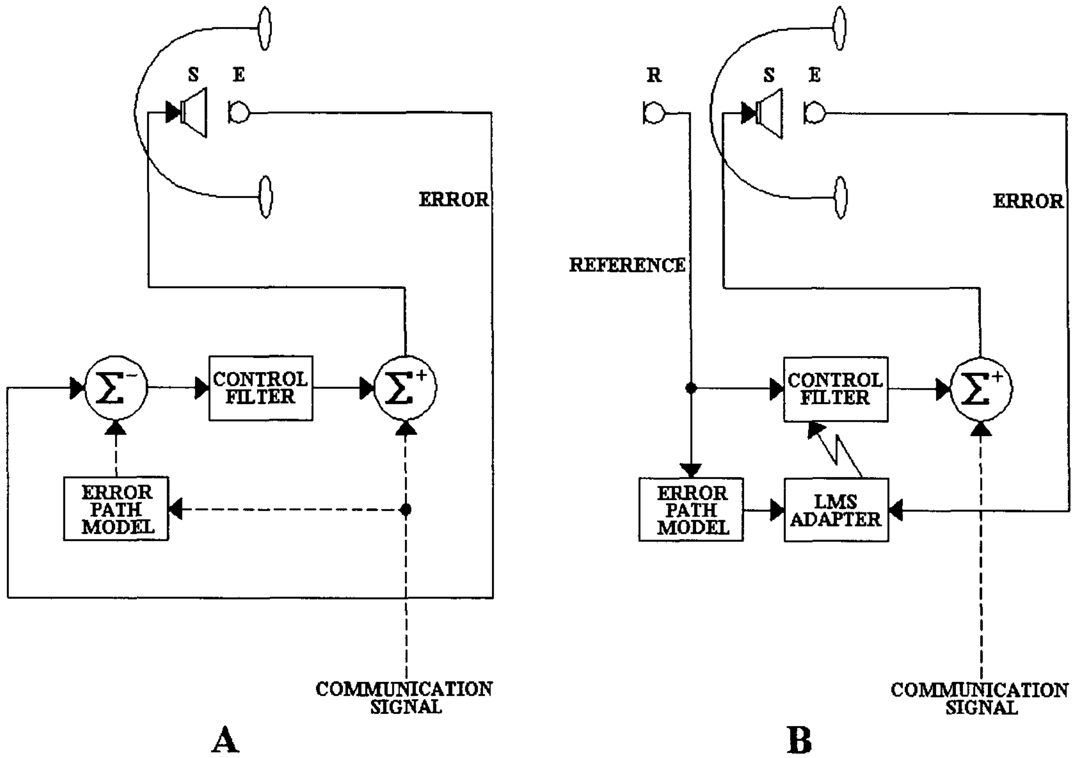
With the laboratories and shops still not fully operational, it was not possible to initiate experimental research during the term of the grant. Instead a number of alternative activities were undertaken.

Interaction between the performance of an active control system and speech communication in headsets equipped with an active noise reduction (ANR) system. A primary expectation of communication headsets is to maintain speech intelligibility under all operational conditions, and especially in circumstances in which a loss of intelligibility may have serious consequences, such as air-traffic control. This requirement may be difficult to maintain in noisy environments and for persons with hearing loss, and also when the communication system is operated at sound levels sufficient to induce temporary threshold shift from speech or spurious electronic signals. The interaction between the performance of the control system and speech communication in ANR headsets has been infrequently examined. The purpose of this work was to analyze this relationship with particular reference to the control structure and, more specifically, the error path.

Analysis of control structure. Simplified block diagrams for one earmuff of a circumaural active headset containing the essential elements of a single-input, single-output, feedback, or feedforward, control system are shown in Figs. 8A and 8B, respectively. A fixed-filter feedback control structure is shown, as it is commonly employed commercially, and an adaptive-filter feedforward control structure that has been applied to a communication headset. The latter has been described by us in detail elsewhere. The complete headset consists of two such earmuffs with cushions to provide an air seal between the earmuff and the head, connected by a sprung headband. Each ear cup contains an independent ANR system. The block diagrams show the signal paths (continuous lines for the control system, and dashed lines for the communication channel) with directions (arrows), and signal summation and subtraction (Σ^+, and Σ^-).

In Fig. 8A, the control filter processes the input signal in a prescribed manner intended to reduce environmental noise at the location of the microphone. An integral part of the process of sound cancellation is the transformation of the electrical signal to sound by the loudspeaker, the propagation of sound from the loudspeaker, S, to the microphone, E, and the transformation of sound into an electrical signal by the microphone. These processes together define the transfer function from S to E, which is termed the *error path*. In essence, the microphone detects the error in the sound produced by the loudspeaker (i.e., the combination of the environmental noise and the canceling sound at E), and hence is known as the *error* microphone. It should be noted that the error path will be significantly influenced by the presence of the head, and any air gap between the cushion of the earmuff and the head (i.e., earmuff "sealed" or "unsealed"), and so is a variable and difficult to control component of the system.

Figure 8: Communication headset with: A – feedback, and B – feedforward active control system, with LMS algorithm to adjust the control filter in B.



The introduction of a communication signal into the control system of Fig. 8A may be performed in several ways. The most desirable control structure is shown in the diagram, which has several equivalent representations. This introduces an approximation for the error path (error path model). The communication signal is directly summed with the signal produced by the control filter at Σ^+ and fed to the loudspeaker. However, in these circumstances the speech sounds are mixed with the environmental sounds sensed by the error microphone, and would be cancelled by the controller unless removed prior to signal processing. This is achieved in Fig. 8A by subtracting the speech signal from the error signal at Σ^-.

The success of this separation between the signal representing the residual noise to be controlled and that representing the speech, and thus ultimately the quality of the speech signal subsequently fed to the loudspeaker, depends on the precision of the error-path model as well as the fidelity of sound reproduction by the loudspeaker. Now the error path is dependent on the subject and the fit of the earmuff to the side of the head. Thus, in a fixed-filter design, it is inevitable that a typical error-path model will be employed, which will be inaccurate in most circumstances and so compromise speech quality.

The same loudspeaker and error microphone are to be found within the earmuff of a headset employing a feedforward control system (Fig. 8B). In this case, however, an additional microphone, R, is used to sense the sound field external to the earmuff. This *reference* microphone provides the input signal to the controller, which must then model the transfer function from the location of the reference microphone to that of the error microphone (i.e., the transmission of sound through the ear cup, as well as leakage between the earmuff cushion and the head, and the transformation of the electrical signal to sound by the loudspeaker). This process is done by successively modifying the filter constructed by the controller, and is implemented digitally by an adaptive filter. The adjustment requires comparing the reference signal, filtered by an error-path model, with the error signal, using an algorithm designed for this purpose. The communication signal is directly summed with the signal produced by the control filter at Σ^{+} and fed to the loudspeaker, as before. Note that in this control structure the output of the error microphone does not enter the control filter, and so cannot influence the speech signal. No compensation for the presence of speech in the error signal is thus required, and no degradation of the speech signal by the control system is expected to occur. The quality and intelligibility of the speech signal may then be expected to depend solely on the fidelity of sound reproduction by the loudspeaker, assuming the electronics introduce insignificant distortion.

Speech intelligibility and ANR. Experiments designed to confirm the extent to which the noise reduction and speech intelligibility of ANR headsets are degraded by the presence of the speech signal have been conducted in other laboratories by our collaborators (Mr. R.B. Crabtree and colleagues). The results confirm the analysis of the extent to which the noise reduction and speech intelligibility of ANR headsets depend on the control structure. The ANR of a headset with a feedforward control system appears to be slightly perturbed by the presence of a speech signal, while the speech intelligibility of this system appears to be significantly greater than that of an ANR headset with a typical feedback control system. The results seem to support the notion that feedforward implementation of ANR is intrinsically less disruptive of communications signals when compared with the more common feedback circuits, and were reported at ICBEN 2003 in the paper: A.J. Brammer, R.B. Crabtree, D.R. Peterson and M.G. Cherniack, "Intelligibility in active communication headsets: Role of Error Path in Active Noise Reduction and Speech Reproduction", Proc 8^{th} Int Congress on

Noise as a Public Health Hazard, Rotterdam (2003) pp. 58-64. The work has also been reported to NATO Task Group TG028, which is studying methods for measuring the performance of ANR communication headsets.

Speech Transmission Index (STI). The STI is an extremely useful physical predictor of speech intelligibility that has been developed over a period of some 30 years, mostly in Europe at the laboratories of TNO in the Netherlands. We took the opportunity to participate in a course given in the Netherlands by the inventors of the STI in order to get rapidly up to speed on current versions of this important metric.

Otoacoustic emissions. Otoacoustic emission has long held promise for the study of hearing mechanisms, but has not generally been found a reliable predictor of acute or persistent hearing loss. Knowledge of hearing acuity is, however, important for maintaining speech intelligibility in a communication system operating in a noisy environment. We took the opportunity to visit an Italian research group in the Dipartimento Igiene del Lavoro, ISPESL, Rome, which appears to have found a method for processing otoacoustic signals so that they correlate better with audiometric measurements of hearing threshold.

Site visits: FAA permission to visit air-traffic control towers is still in the approval process, and the visits are expected to be undertaken in 2004.

Staff development: Mr. S. Gullapali, who was trained on electronic printed circuit board design and fabrication methods for this project, will be maintained for the next 12 months on internal funds to further develop methodology pertinent to air-traffic control communications in noisy environments.

A.J. Brammer
February 4, 2004