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DIGITAL CORRECTION FOR EXCESSIVE AIR ABSORPTION IN ACOUSTIC SCALE MODELS

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1. INTRODUCTION

The main objective of the research to which this paper relates is to investigate the effect of the scattering parameters of fittings on sound fields in industrial spaces. It has been established that acoustic conditions in disproportionate industrial spaces can be characterised by the parameter known as sound propagation [1,2]. Results obtained by the authors from computer simulation in previous work have shown that the sound propagation characteristics of disproportionate rooms are determined by the product of scattering parameters (scattering cross section, Sx, and absorption coefficient, αobst) of fittings [2,3]. It follows that measurement of sound propagation in a disproportionate room with various shapes of fittings and distributions inside the room could be used to determine the value of this product. In the previous work it was also suggested that the value of the product of those two parameters might be related to the absorption as measured in a conventional reverberation chamber. Therefore, it is proposed that measurements be carried out in both disproportionate chamber and reverberation chamber to seek relationships between these two parameters.

To obtain the necessary amount of systematic data, numerous tests will have to be carried out. It would be very difficult to employ real chambers (full scale chambers). The alternative to the use of full scale test facilities is physical scale modelling. However, the most serious problem encountered when working with scale models is to scale down all the relevant acoustic parameters especially air absorption and boundary absorption. Air absorption consists of classical absorption and molecular absorption which increase as the square of frequency. Therefore scaling frequency by a factor of K will increase the air absorption by a factor of K^2.

2. METHODS OF SOLVING THE PROBLEM OF THE EXCESS AIR ABSORPTION

There are two approaches to minimise the excess air absorption in acoustic scale models: these are physical methods and numerical methods. Physical methods, known as traditional methods, employ one of two techniques to reduce the relative humidity of air inside a model chamber since the air absorption at low values of relative humidity is small at model frequency.

The first is by removing the water vapour with the technique of circulating dry air or by removing oxygen with the use of nitrogen in the model. Removing oxygen will affect the speed of sound in the model. A numerical method was proposed by Polack et al [4] in 1985 which employed a computer technique to compensate for air attenuation. The algorithm for the calculation of sound decay for octave bands of noise developed by Polack is based upon the Fast Fourier Transform (FFT) technique. Recently, Polack [5] reported the successful application of his system (called MIDAS) to small models over a limited frequency range.

The numerical method appears to be more convenient than the physical method since measurements in the scale model can be made under ambient conditions. It was for this reason that the authors decided to apply a numerical method to correct for excessive air attenuation in our proposed scale model experiment.

In this paper a new algorithm for the numerical compensation for air absorption in acoustic scale models is proposed. Unlike Polack's method which is based upon the FFT algorithm, a digital filtering algorithm is employed. A numerical compensation method similar to that employed in the Polack technique was developed to facilitate comparison of results obtained by using the digital filtering algorithm with results obtained using the FFT algorithm. It was found that slopes of energy decay calculated by those two algorithms were in a good agreement but that the digital filtering algorithm was much faster.

3. NUMERICAL COMPENSATION OF AIR ATTENUATION

The basic formula used for predicting the sound attenuation in air has been derived in detail by Bass et all [6], and can be found in an American Standard [7]. It is given as:

$$\alpha=4.34(2\pi f/c)^2[l_v+(\gamma-1)l_h],\ dB/m$$

l_v and l_h are characteristic values of lengths for viscosity and for heat given by

$$l_v=(4/3\mu+\eta)/\rho c,\ l_v'=\mu/\rho c,$$

$$l_h=\lambda/\rho c C_p$$

where

- α is the absorption coefficient of the air
- f is the frequency
- c is the speed of sound
- ρ is the density
- μ is the shear viscosity
- η is the bulk viscosity
- λ is the coefficient of thermal conductivity
- γ is the ratio of coefficient for the specific heat at constant pressure and at constant volume (C_p/C_v)

The values of μ, η, λ and ρc are depend on pressure and temperature of the air. Only the value of η is function of frequency and is dependent on the relative humidity of the air.

Since the speed of sound is independent of frequency, it means that travel distances are simply related to travel times. This if an impulse source is employed the time of arrival of a particular reflection can be used to calculate the distance that the sound has travelled. Therefore compensation for air attenuation can be achieved for a particular frequency as folows:

$$E_1 = (1/e^{-\alpha ct})\, E_o$$

where

- E_o is the energy without air compensation
- E_1 is the energy with air compensation

3.1 FFT Based Compensation Technique

The principle of decay computation based upon the FFT technique is illustrated in Figure 1(A). The spectrum of a time windowed portion of the signal is computed by FFT at regular time intervals corresponding to the movement of the time window along the total time history. Two important parameters must be specified properly, namely the length of the window and the window movement between successive calculations. The length of window is determined by the number of samples required for the FFT, while the window increment is specified by the amount of overlap of adjacent windows. At each of the spectral lines calculated by the FFT the appropriate correction can be made for air absorption.

3.2 Digital Filtering Based Compensation Technique

Principle of decay computation based upon the digital filtering technique is illustrated in Figure 1(B). Since the time history is actually a digital signal it can be filtered using a digital filter covering a particular frequency band. The digital filters employed in this work were designed using MATLAB software. There are two types of filters which are commonly used in a digital signal processing: Finite Impulse Response (FIR) filters and Infinite Impulse Response (IIR) filters. Two MATLAB functions can be used to perform direct filter designs using either the Yule-Walker method for an IIR filter and the Parks-McClellan method for a FIR filter. At the present time the Yule-Walker approach has been employed since the MATLAB function enables the design of an arbitrarily shaped multiband frequency response filter to be carried out with relative ease. However, it is intended that in the near future a comparison of the performances of two different filter designs will be made.

The filtered signal is squared to become a power time series, then the power time series is grouped into successive 10 msec bins in order to smooth the data. The contents of each bin is actually the summation of power of each component of the power time history that falls within that bin. At this stage the air absorption compensation is implemented and then the logarithm of the total power in the bin is calculated.

4. SIMULATION OF IMPULSE RESPONSE

Numerical methods for the compensation of air attenuation must employ an impulse source. It was decided to utilize a computer simulation of a sound source before embarking upon actual measurements. The impulse response of a room was simulated by use of the simple image method of Gibbs and Jones [8] rather than use a ray-tracing method. This particular model has been chosen since (i) the model can be use to predict sound pressure level throughout a rectangular room which could be similar to the characteristics of a disproportionate room, (ii) the model can be run on the computer efficiently and rapidly. Simulations were made for a rectangular room with an equivalent full-scale dimensions of 10mx8mx3m.

One objective of simulation was to investigate what resolution of an analog to digital converter (ADC) would be needed to work up to a particular maximum frequency in a scale model. This is very important because the quality and accuracy of the data acquisition system of the measurements rely heavily on the ADC which must satisfy two important requirements: Firstly, the dynamic range of the decay has to be large enough (at least 20 dB) in order to calculate a reverberation time correctly. Secondly, the conversion rate must be adequate in order to encompass a maximum test frequency. It should be mentioned that the cost of ADC systems is very dependant on the resolution and conversion rate. In the author's earlier work, it was suggested that scale factor of 20 would be appropriate for use in our experiments. Simulations have thus been made for a model at scale 1:20. Regarding the resolution needed for a scale model, Polack [4] reported that the MIDAS system (14 bit ADC with a sampling frequency up to 333KHz) can fulfil easily the two important requirements above for a scale factor of 1:50 with a model frequency of up to 160kHz. Compared to Polack's scale model, the requirements of our proposed model are less severe. A data acquisition system with the same specification as the MIDAS system would definitely satisfy the above requirements on scale model of 1:20. However, the simulation enables an examination to be made of the possibility of using a low cost ADC with a resolution of 12 bits in our model.

In addition, the simulation can also help to determine some parameters specified in the calculation. For example, with the FFT algorithm the number of samples for the FFT and the amount of overlap must be optimised, while with the digital filtering algorithm the digital filter must be designed as accurately as possible.

5. DISCUSSION OF SIMULATION RESULTS

Simulations were made for both 12 and 14 bit resolutions of the ADC for equivalent full-scale frequencies of 500Hz, 1kHz and 2kHz. A sampling rate 200KHz was employed. The slopes of the octave band decays obtained using the digital filtering algorithm and those obtained using the FFT algorithm were very similar. The effect of compensation appears significant only in the 2kHz octave band. It is therefore the graphs which show the dynamic range of the sound decay with air absorption compensation at 2kHz which should be used to examine the ADC resolution needed in a model with a scale of 1:20. As expected, an ADC with 14 bit resolution can fulfil the requirements perfectly as shown in Figure 9 the dynamic range is about 40dB. While with a 12 bit resolution, the dynamic range that can be achieved is about 20dB as shown in Figure 7. This suggests that an ADC with 12 bit resolution may be suitable for use our proposed experiments since a minimum dynamic range of 20dB is just acceptable. It can be observed that the dynamic range achieved is slightly better for the digital filtering algorithm than for the FFT algorithm.

Table 1 shows the results of Reverberation Time (RT) determined by both algorithms with 12 and 14 bit ADC resolution and calculated using the Sabine theory. All results were calculated from the decay curve with compensation for air attenuation. The RT obtained for both ADC resolutions for each particular frequency are generally similar. When comparing the numerical results with the RT calculated by Sabine theory, the differences were small being greatest at frequencies of 500Hz and 1000Hz. This suggests that both algorithms for the numerical compensation of air attenuation could be employed for processing data from experiments with confidence.

Table 1 RT results from Simulation

Frequency	Sabine	Digital Filtering method		FFT method	
		12bit	14bit	12bit	14bit
	1.432				
500		1.38	1.38	1.26	1.56
1000		1.35	1.35	1.32	1.62
2000		1.48	1.41	1.47	1.44

6. CONCLUSIONS

Two algorithms (Digital Filtering and FFT) for calculating compensation for air attenuation in scale models have been developed successfully with the use of MATLAB software. The slopes of energy decay obtained by both algorithms very similar. However, it is suggested that the use of the digital filtering algorithm could be better approach to air absorption compensation since with this algorithm the speed of calculation is much faster.

From the simulation of the impulse response for the test frequency maximum equivalent to full-scale value of 2kHz, it is suggested that the data acquisition system for a model with scale of 1:20 an ADC with 12 bit resolution would be adequate.

7.REFERENCES

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[8] B.M. Gibbs and D.K. Jones, "A Simple Image Method for Calculating the Distribution of Sound Pressure Levels within an Enclosures", Acustica 26, 1972, pp.24-32.

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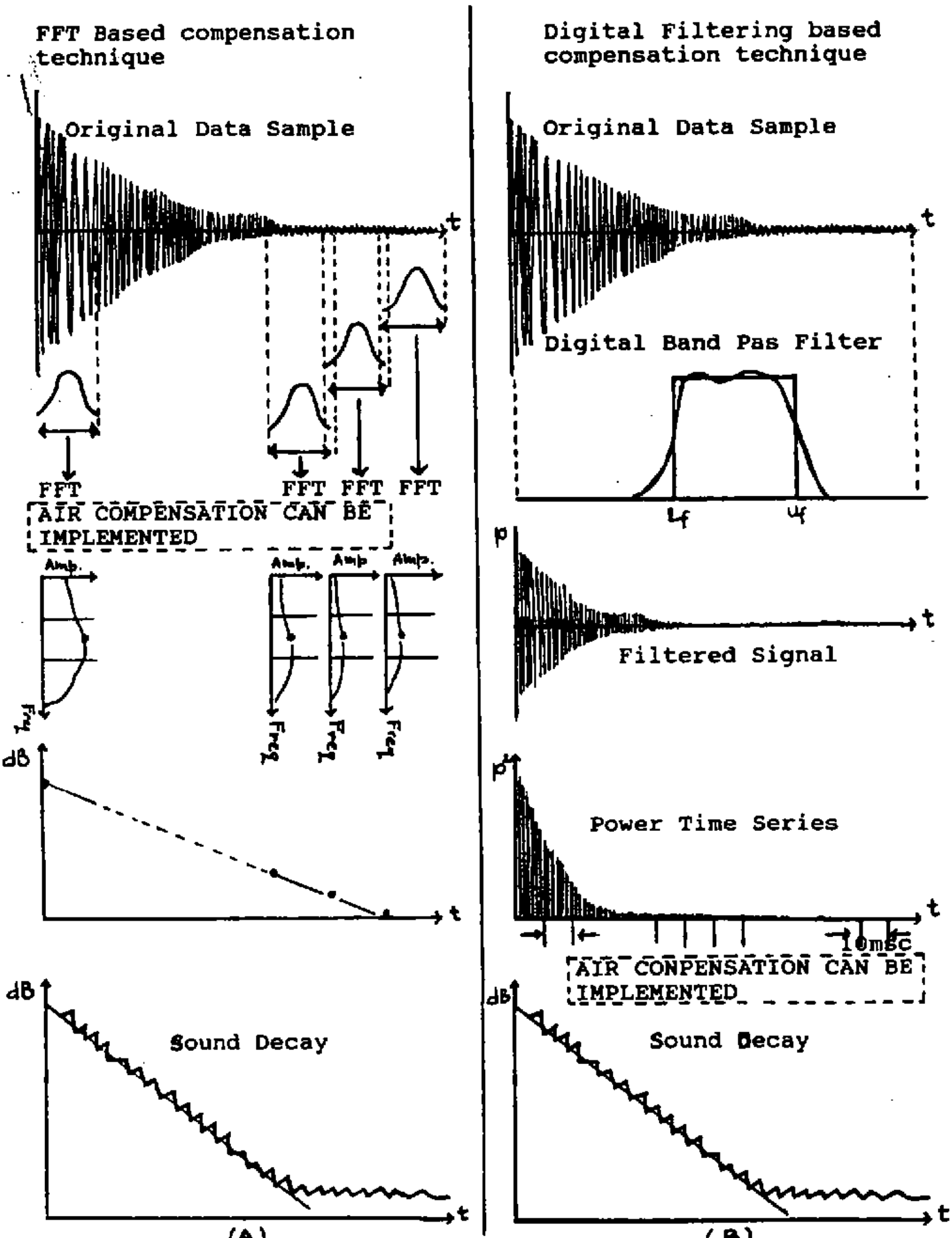

Figure 1 Principle of Decay Computation for compensation of air attenuation

DIGITAL CORRECTION

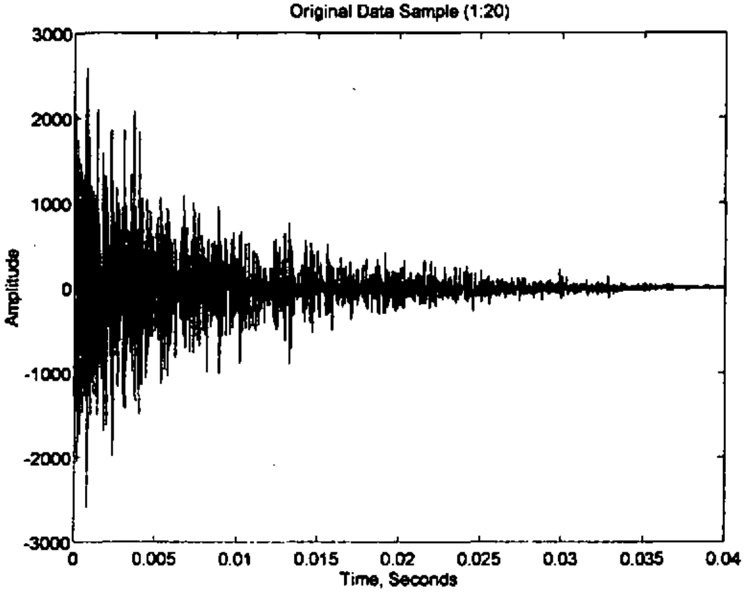

Figure 2 Original Data Sample with 12 bit resolution of ADC

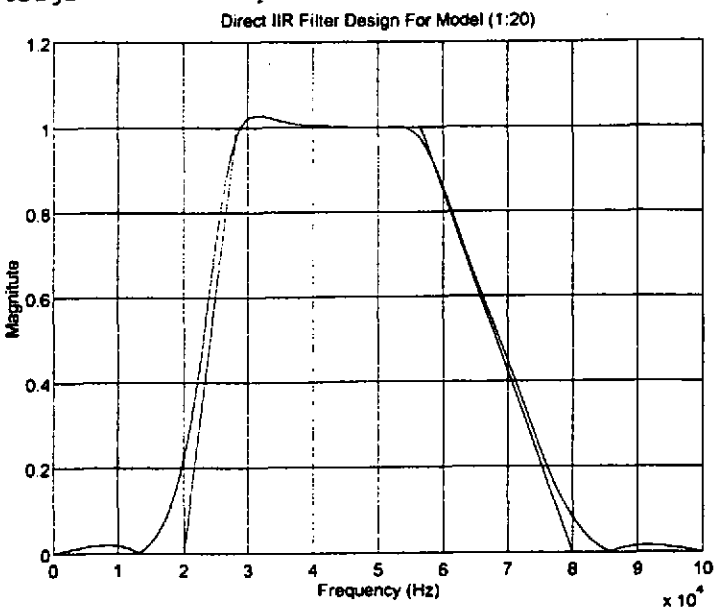

Figure 3 Octave Band Filter equivalent full scale of 2kHz

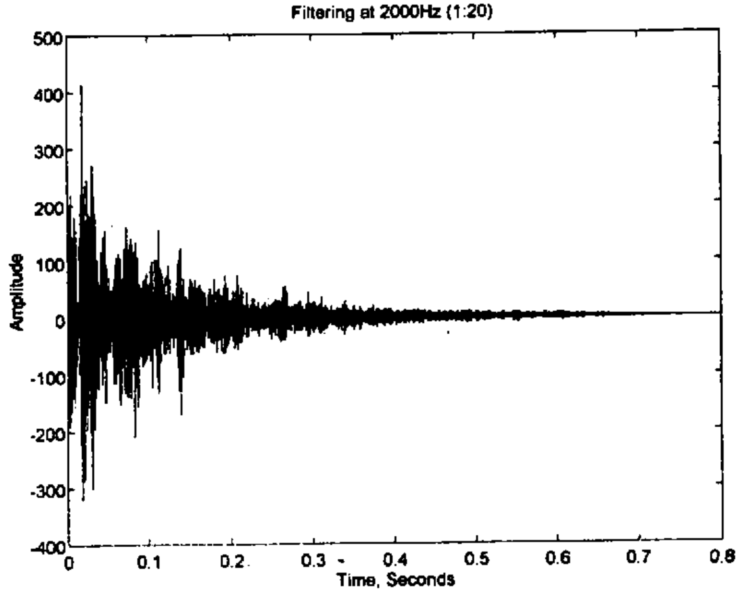

Figure 4 Filtered Signal (2KHz) of Original Data Sample above

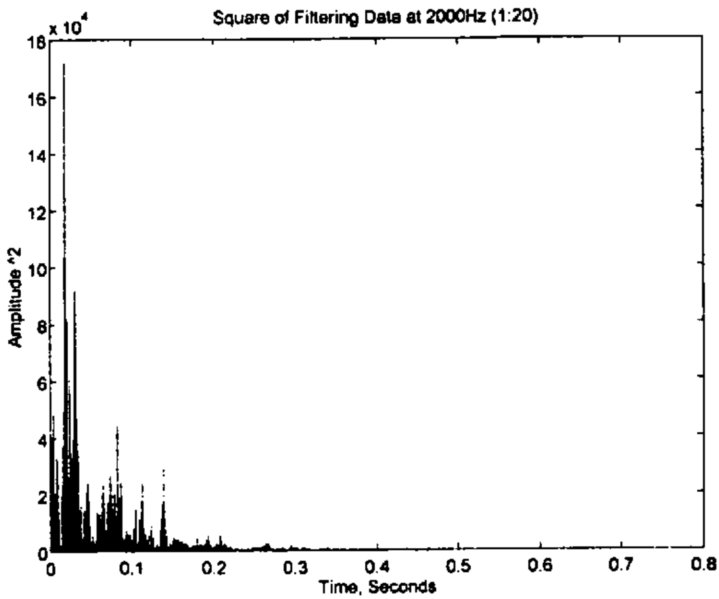

Figure 5 Power time series of filtered signal at 2kHz

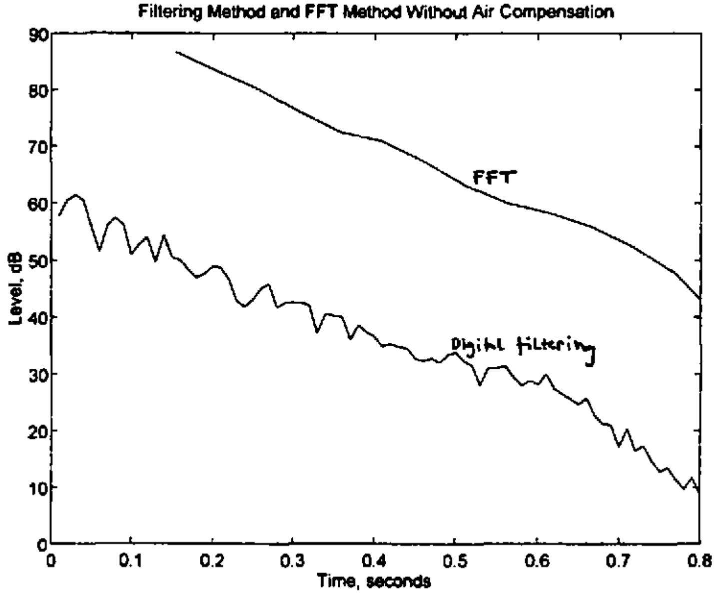

Figure 6 Sound Decay of Original Data (12 bit resolution)

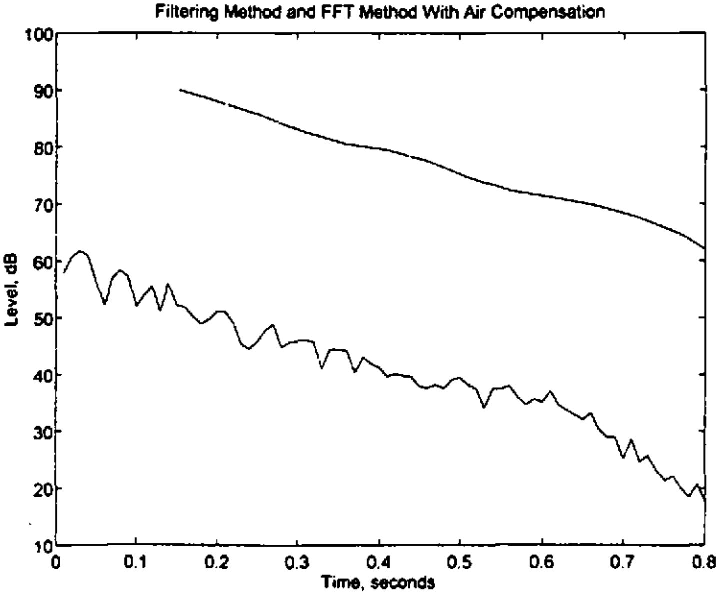

Figure 7 Sound Decay of Original Data (12 bit resolution)

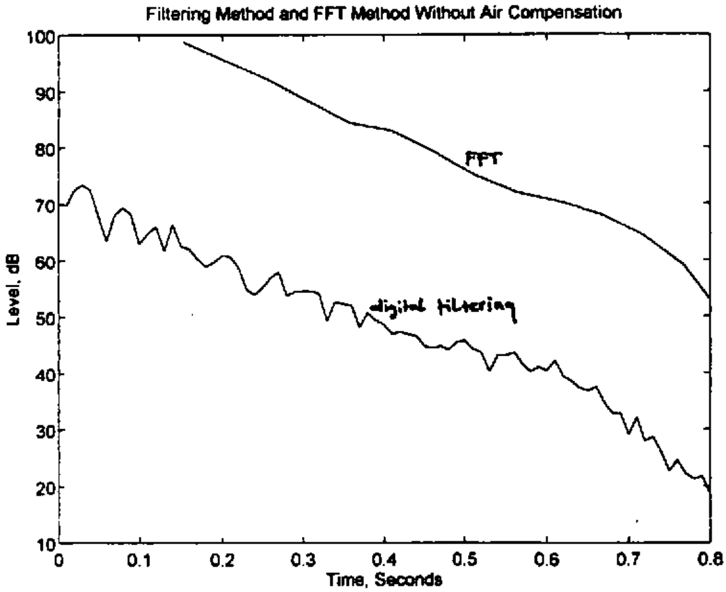

Figure 8 Sound Decay (14 bit resolution)

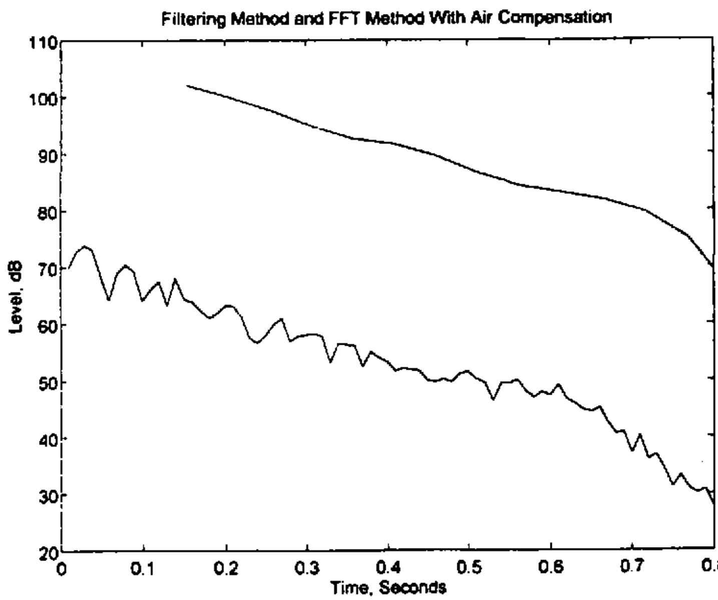

Figure 9 Sound Decay (14 bit resolution)