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SMART SURFACES FOR BUILDING ACOUSTICS

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1. INTRODUCTION

The acoustics of an architectural space are defined by its geometry and the impedance of its boundaries. The impedance presented to acoustic waves at each boundary is determined by the properties of the construction materials. This impedance controls the reflection of acoustic waves and the absorption of incident energy and so dictates the modal characteristics of the space.

Conventional noise control can change the impedance of a boundary with passive absorbers. At higher frequencies single layer absorptive materials work well, whilst at lower frequencies the required depth of material becomes impractical. Tuned resonators are then used with limited bandwidth. Arrays of tuned resonators are necessary for broad low frequency absorption with complex design and equipping requirements.

An alternative is to use "active" or "smart" boundaries. It is possible to modify the dynamics of the boundary and so influence the impedance presented to acoustic waves using active control. For a simple compliantly suspended boundary with appropriate adaptive control the impedance can be precisely specified with any desired value limited by the forcing actuator capabilities. For example, perfectly anechoic or reflective boundaries can be selected by software control.

The required physical volume is minimal compared with passive low frequency absorber design. The programmable nature of such a device would facilitate easy on-site tuning to optimise a particular space. Other applications of the programmable impedance concept include acoustic test loads, especially where flexibility and small size are important.

2. ACTIVE CONTROL OF ACOUSTIC IMPEDANCE

2.1 Theory

Consider an rigid infinite boundary mounted on a simple linear suspension. If there are normally incident plane waves with consistent pressure over the boundary it is possible to determine the acoustic impedance by isolating a unit area. Figure 1a shows the mechanical components of such a system.

The equation of motion of the boundary plane is:

$$M\frac{d^2x}{dt^2} + R\frac{dx}{dt} + Kx - p = 0 \qquad (1)$$

M is the mass per unit area, K the stiffness per unit area of the suspension and R is the resistance to the motion. The displacement of the boundary is x and the pressure at the surface is p.

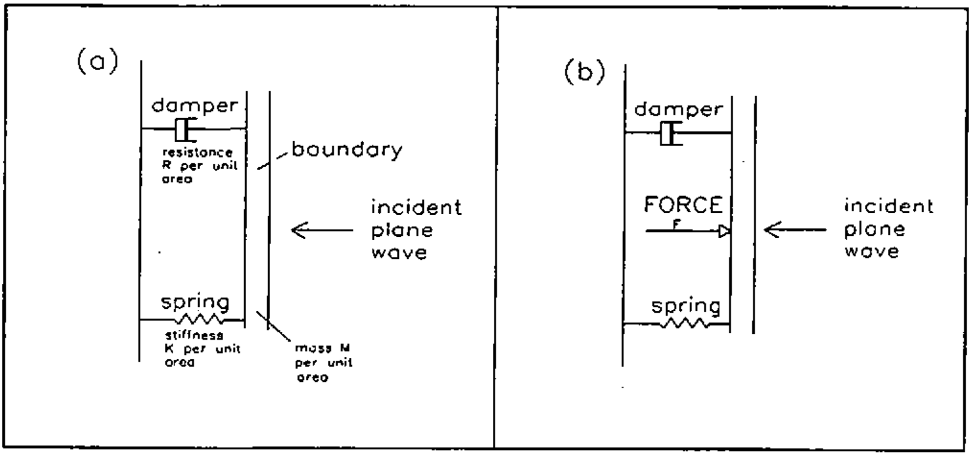

Figure 1. Compliantly suspended boundary without (a) and (b) with forced input.

By taking a Fourier transform and rearranging, the specific acoustic impedance z at the surface of the boundary can be specified in terms of the particle speed u and frequency ω:

$$\frac{p}{u} = \mathbf{z} = R + j\left(M\omega - \frac{K}{\omega}\right) \qquad kgm^{-2}s^{-1} \qquad (2)$$

An example is displayed in Figure 2, in which a loudspeaker is used as the boundary element. Active control can be introduced by applying a force input at the boundary (see Figure 1b). The equation of motion is then:

$$M\frac{d^2x}{dt^2} + R\frac{dx}{dt} + Kx - p = F \qquad (3)$$

where F represents the forcing function. This can be expressed as (4) by Fourier transform and rearrangement.

$$\frac{p}{u} = \frac{F}{u} + R + j\left(M\omega - \frac{K}{\omega}\right) \quad kgm^{-2}s^{-1} \quad (4)$$

Note the presence of the variable u on both sides of the equation. If a control system can create the appropriate forcing function for p and u then the acoustic impedance at the surface can be altered to any desired value.

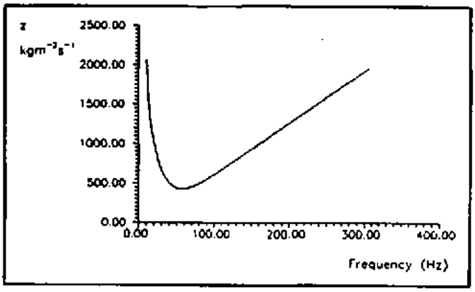

Figure 2. Predicted input specific acoustic impedance of KEF B200A loudspeaker when connected to a power amplifier.

2.2 Signal Processing Requirements

Previous research has used analogue control to vary the surface acoustic impedance. For a steady state harmonic signal, control can be implemented by filtering the source signal with manually adjusted gain and phase characteristics to create a forcing signal [1,2,3].

This research uses a different approach. Typical "real" signals are not steady state and may be quasi-periodic or random; adaptive digital control must then be used. The linear adaptive digital filter topology known as the Least Mean Squares (LMS) algorithm [4] is suitable; this and other LMS variants have been succesfully applied in numerous noise and other control problems.

The required performance of the control system also depends on the linearity of the forcing system and the boundary. The boundary may exhibit a non-linear relationship between the applied force and displacement; non-linear adaptive controllers are then necessary.

3. ONE-DIMENSIONAL PRACTICAL IMPEDANCE CONTROL

Although this research is primarily motivated to identify the non-linear adaptive digital algorithms required to control the acoustic impedance of plates and partitions, the starting point is the linear adaptive control of a simple compliantly suspended loudspeaker operating in its' piston (or first-mode) range.

A typical quality loudspeaker (such as the KEF B200A 8inch l/s used here) operating at lower levels has a linear characteristic between the input (or forcing) signal and the piston displacement. This facilitates the use of linear algorithms such as the LMS method and variants.

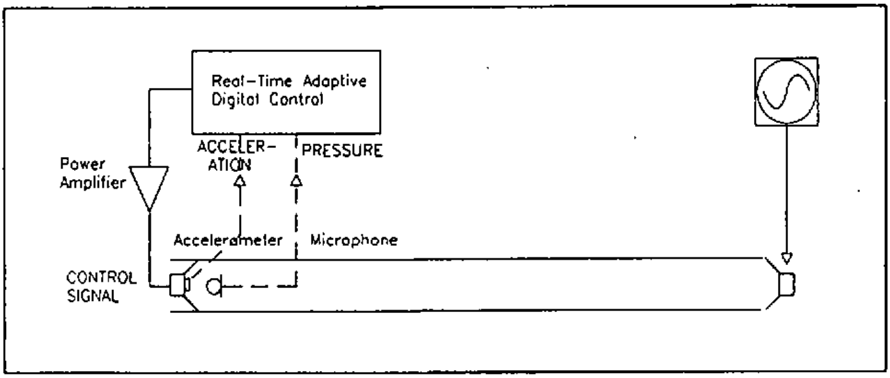

Figure 3. One-dimensional acoustic impedance test rig.

An outline of the test rig is displayed in Figure 3. The noise source loudspeaker is on the right, with the controlled-impedance piston on the left. Both loudspeakers are mounted in sealed boxes. The loudspeakers operate in first-mode which restricts the maximum source frequency to 700Hz.

The system measures the acceleration of the cone, and the pressure at the surface. The two measured signals are used by the control system to generate a signal that is fed to the voice-coil. This control signal changes the motion of the cone, so changing the acoustic impedance. The microphone is selected to have low sensitivity to lateral body movement, and is mounted on the piston. The output of the model used contains acceleration components of >38dB below pressure components. This measurement system is distinctly different from other research efforts [1,2,3,5], which sense pressure at locations away from the boundary.

The control system uses a recursive linear combiner filter (Figure 4) to generate the control signal. By changing each tap's gain different transfer functions are created. The filter is made adaptive by updating the weights with a variation on the LMS algorithm known as the Recursive Least Mean Squares (RLMS) method [3,6]. This algorithm in conjunction with the combiner creates a recursive adaptive filter with an infinite impulse response.

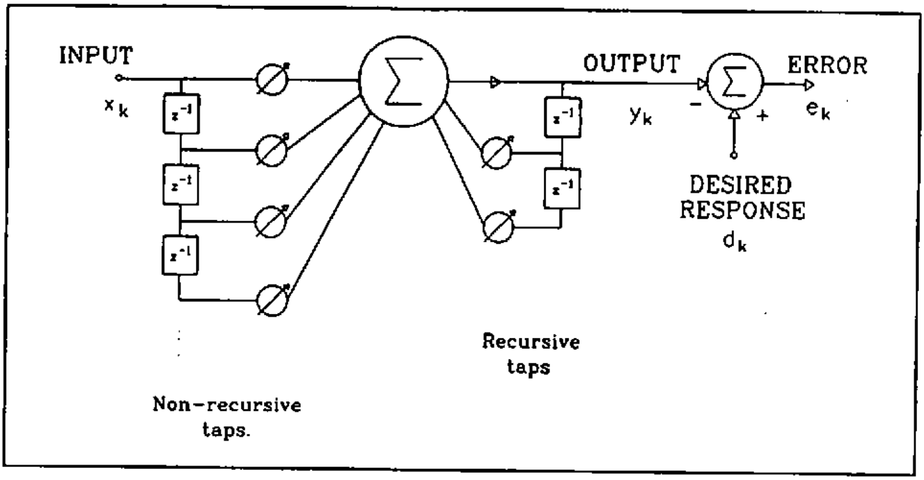

Figure 4. The linear combiner as a recursive transversal filter.

The control system topology is displayed in Figure 5.

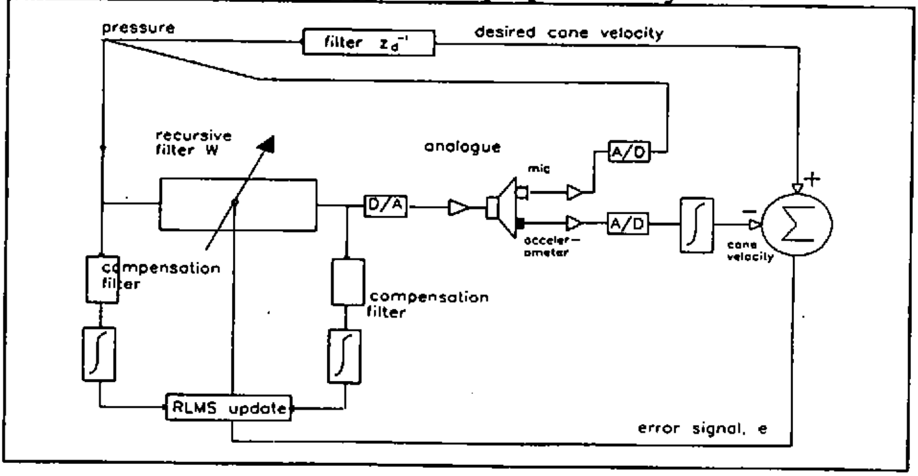

Figure 5. Controlled impedance adaptive system topology

The input to the filter is the measured cone pressure. The filter output is the control signal and is fed to the speaker voice-coil. The desired velocity signal is a filtered version of the pressure signal, this filter specifies the desired impedance. The desired velocity signal is compared with the integrated acceleration of the cone (the actual velocity) and the error is used by the RLMS algorithm to update the recursive filter coefficients. The control system is implemented on a Loughborough Sound Images system board that contains an AT&T DSP32C processor.

The adaptive filter topology uses the filtered-U algorithm [6]. The transfer function between the filter output and the input to the error summation must be estimated, then implemented in the compensating filters shown. Without this compensation the adaptive control will not be stable. The test rig estimates the transfer function off-line. Pulses are sent from the DSP through this forward path, and sampled. Coefficients for a time domain compensating FIR filter are then determined.

4. RESULTS

The surface of the B200A speaker was subjected to a harmonic wave of frequency 125Hz, with a sound pressure level of 110dB at the surface of the cone. The effect of the controller on the surface pressure (p) and velocity (u) for different desired acoustic impedances is shown in Figure 6.

Without control there is a phase difference at this particular frequency between p and u. For ideal absorption the desired impedance filter is programmed to create $z = 415$ at the cone. The test results show that p and u are then in phase, and their magnitudes have the correct ratios.

The acoustic impedance can be made very large ($u \to 0$); however there is a practical limit imposed by the noise of the accelerometer signal, especially at lower frequencies where integration causes large gain.

If a desired acoustic impedance is particularly low ($p \to 0$), then the control topology of Figure 5 will not have enough input signal for the adaptive filter. The control topology has to be rearranged so that u is input to the filter. Low acoustic impedance can then be obtained as shown in Figure 6. The microphone noise floor imposes a lower limit on z.

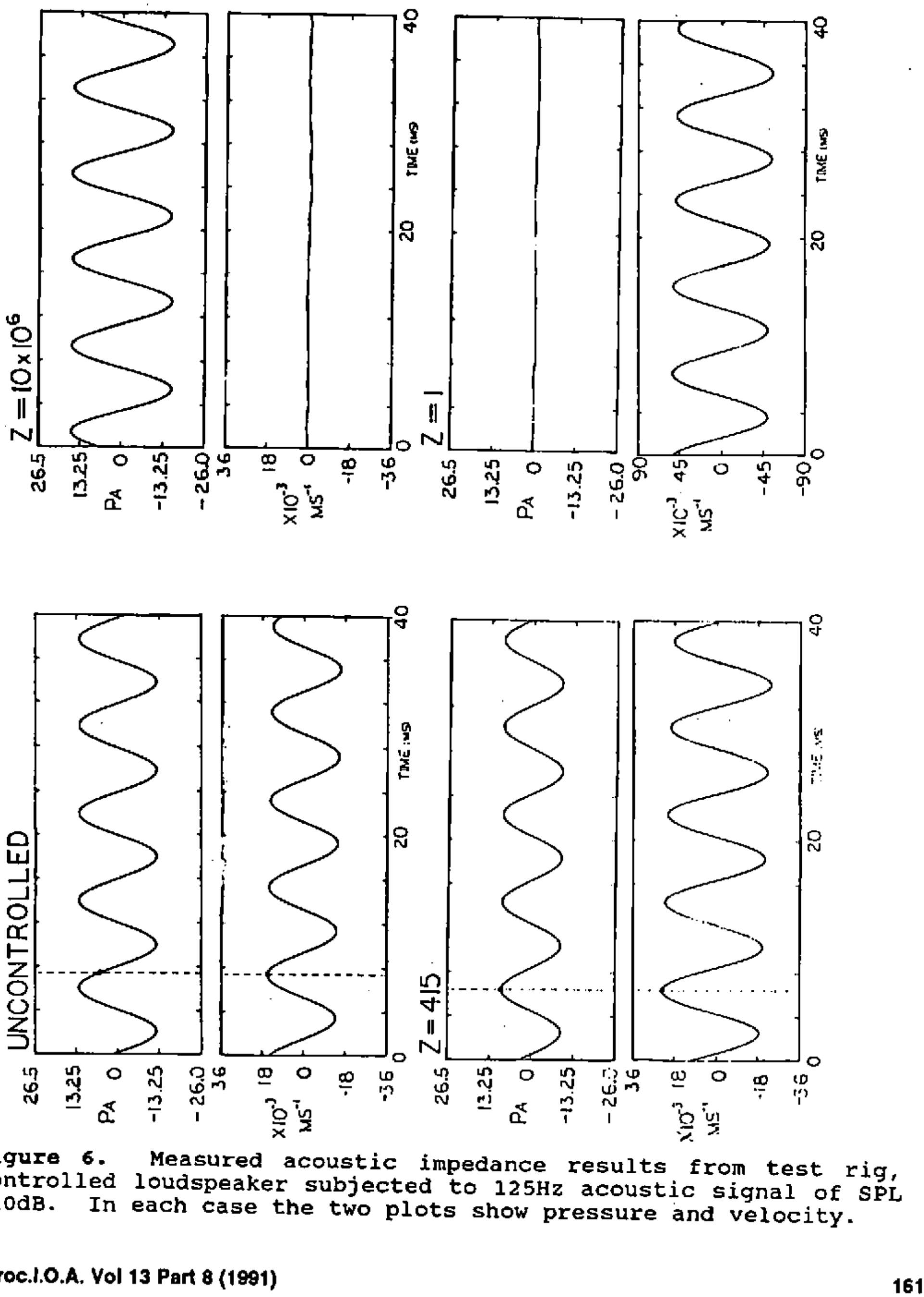

Figure 6. Measured acoustic impedance results from test rig, controlled loudspeaker subjected to 125Hz acoustic signal of SPL 110dB. In each case the two plots show pressure and velocity.

4. CONSTRAINTS

The system has a number of constraints. Firstly there is the physical limitations incurred by the maximum loudspeaker excursion and electrical power limits. Secondly the measurement system creates a fixed feedback path around the adaptive filter, which tends to reduce the stability margin of the update process.

4.1 Loudspeaker Constraints

The conventional moving-coil loudspeaker is limited by the maximum linear displacement the cone/suspension can achieve and the maximum electrical power that the coil can dissipate.

The maximum displacement and frequency defines the cone velocity amplitude which for the KEF B200A loudspeaker is

$$|u_{max}(\omega)| = 3.15\times10^{-3}\,\omega \quad \frac{m}{s} \tag{5}$$

The electrical power constraints on cone velocity can be established by substituting the equation of motion at the surface of the cone (see (4); note that pressure must be multiplied by area) with the force provided by the voice-coil:

$$F = \frac{BlV - (Bl)^2 u}{Z_{EB}} \tag{6}$$

where V is the voltage supplied to the voice-coil and Z_{EB} is the blocked electrical impedance. Bl is the force factor of the drive system and u is the cone velocity. Substituting (6) into (4) and rearranging to express the variables in terms of velocity gives (7). There are four variables : voltage across the voice-coil, cone velocity, the surface impedance **z** and frequency. If a control system can achieve a desired **z** then the maximum voltage and frequency will define the maximum cone velocity u.

$$u_{max} = \frac{Bl\,V_{max}}{Z_{EB}}\left[\left(S\mathbf{z} + \frac{(Bl)^2}{Z_{EB}} - R\right) - j\left(M\omega - \frac{K}{\omega}\right)\right]^{-1} \tag{7}$$

Using these equations the maximum sound pressure at the cone for a desired impedance can be defined (an example is given in Figure 7). The SPL limits are defined at lower frequencies by (5), and at higher frequencies by (7). The force-displacement of the KEF B200A at these extremes is likely to be non-linear. It is expected that non-linear adaptive filters would be required to control this loudspeaker at such pressure levels.

4.2 Feedback around the Adaptive Filter

A further constraint is imposed by the measurement system which causes a feedback path (Figure 8). The mathematics of convergence of an adaptive system in the presence of feedback are intractable. Effort in control has generally tried to cancel feedback paths using fixed compensation filters [7]. Recursive adaptive algorithms have been shown to offer more stable adaption when operating in the presence of feedback [6]. The authors' experimental observation of controlled-impedance convergence with non-recursive and recursive LMS filters found recursive structures to be more stable; hence the system described in this paper uses a recursive adaptive filter structure.

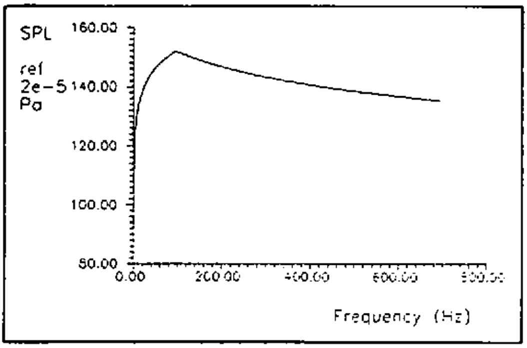

Figure 7. Predicted sound pressure limit at the surface of the cone imposed by power and displacement limits for the KEF B200A with a controlled impedance of $\rho_0 c$

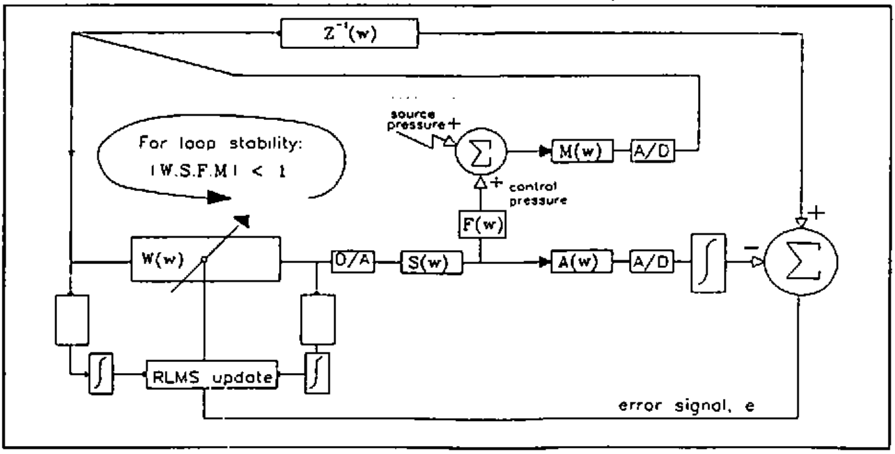

Figure 8. Feedback path around the adaptive filter

The analysis of stable converged solutions provides insight to system set-up. If the system is to converge then the error e

$$W = \frac{\omega z_d^{-1}}{z_d^{-1} MSF + SA} \quad (8)$$

$$|A| > |\omega z_d^{-1} MF| \quad (9)$$

must go to zero, transfer function analysis gives (8). Substituting the loop stability condition defined in Figure 8 with (8) gives an expression (9) that must be satisfied for stable converged solutions. If no feedback path exists (ie: F=0), then converged solutions are stable. If F has non-zero values, stability is achieved by placing gain and attenuation in the digital and analogue domains to satisfy (9). Note that (9) does not guarantee system convergence, because there is no account of system dynamics.

5. CONCLUDING REMARKS

The control of the acoustic impedance of a piston operating linearly in first-mode has been demonstrated. Further research will establish more stable control for the feedback stability problem. The control of plates and partitions will require non-linear adaptive systems, these are currently being investigated.

6. REFERENCES

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