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RECENT RESEARCH TOPICS IN SOUND FIELD CONTROL AT NTT LABORATORIES

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1. INTRODUCTION

Recent progress in digital signal processing is bringing acoustics into modern science. Active noise control and sound field control are typical examples of new topics[1]. Sound field control is mainly based on adaptation and optimization algorithms which are very useful in signal processing. To date, room acoustics have been designed using a closed-form formula obtained analytically or approximately even when the acoustic system was too complex for it. Sound field control, however, will be a new approach, based on computer science, to complicated room acoustics.

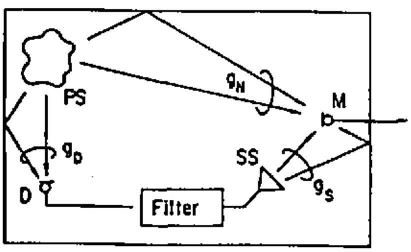
Nippon Telegraph and Telephone Corporation (NTT) is now also doing research on acoustics and audio-technology. This paper briefly reviews on-going research in NTT Human Interface Laboratories on sound field control in a reverberant space and an echo-canceller for a tele-conferencing system. Fundamental studies on transfer functions in a reverberant space are also described. These theoretical investigations will provide useful results about room acoustics for sophisticated sound field control.

2. SOUND FIELD CONTROL IN A REVERBERANT SPACE

Active noise control is an important example of sound field control problems. Figure 1 illustrates a typical condition for active noise control using a secondary source SS. There are two approaches to active noise control: one is to make a specific point M quiet, and the other is to reduce the total radiated power from the noise source PS. These approaches are called quiet-zone control, and active power minimization [2][3].

2.1 Quiet-zone Control Based on MINT

The filter in Figure 1 generates a noise-cancelling signal to suppress the noise level from PS at a point M. The noise cancelling signal is synthesized from the primary source noise signal received by a detection microphone D. This primitive configuration, however, does not work well when the detection microphone D cannot be set close to PS or when SS cannot be set close to the noise-control point M. This is because the detected noise signal or controlling sound from SS will be distorted by reverberation.



PS : Primary sound (noise) source
SS : Secondary source

Fig.1
Sound field control in a reverberant space

Those distortion effects can not be removed by usual inverse filtering, because a transfer function (TF) between two points generally has non-minimum phase zeros in a reverberant space[4][5][6].

Figure 2 shows a schematic diagram that enables inverse-filtering of TFs according to MINT (Multiple input/output INverse filtering Theorem)[7]. Two secondary sources SS1 and SS2, and two filters h_1 and h_2 are used to control the sound pressure response at a point M. Inverse filtering requires that the filter's input signal $x(n)$ and the output (observed at point M) signal $x(n-\tau)$ are identical except for the time delay τ. Thus, the filters h_1 and h_2 must satisfy the equation

$$h_1(z)g_{s1}(z) + h_2(z)g_{s2}(z) = z^{-\tau} \qquad (1)$$

where z denotes the discrete complex frequency, g_{s1} and g_{s2} are the TFs, and τ is the sampled delay time. If we assume here that these TFs are sufficiently long but have records with finite duration, and that the TFs do not have any common zero in the z-plane, then there exists a pair of stable FIR filters, h_1 and h_2. We call this algorithm MINT. MINT can be applied to a system for active noise control in a reverberant space [8][9]. The sound pressures at N points can be controlled using $N+1$ noise-control units. Each control unit is composed of a microphone for detecting noise (the primary source signal), a filter, and a secondary source.

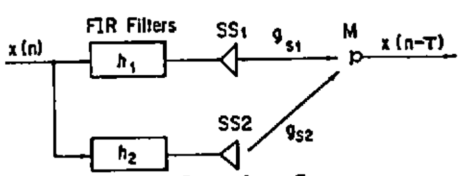


Fig.2 Inverse filtering of transfer functions

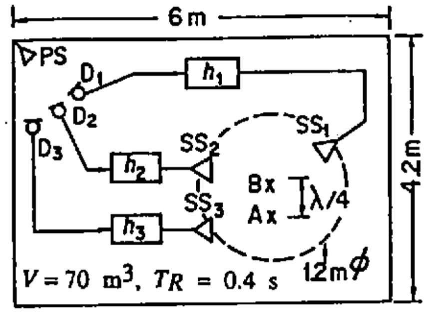


Fig. 3a Experimental configuration

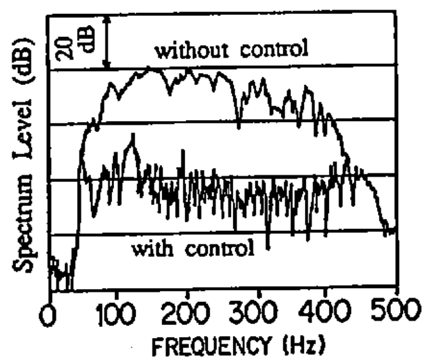
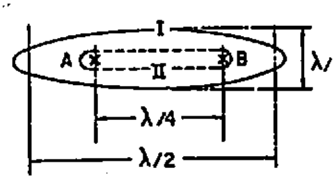


Fig.3b Power spectrum of suppressed noise at point A



I: suppressed by over 6 dB

II: over 14.5 dB

Fig.3c Quiet-zone created around the points A and B

Fig.3 Quiet-zone control in a reverberant room
λ = 152 cm (wave length of the center frequency of the noise from 50 to 400 Hz)

Figures 3a,b,c show experimental results for creating a quiet zone around two noise-controlled-points A and B in a reverberant space. The noise at point A is reduced by more than 25 dB over a relatively wide frequency range when the noise signal extends from 50 Hz to 400Hz (Fig. 3b). Figure 3c illustrates the quiet zone created around the two points where the noise level reduction is more than 6 dB. Studies of such quiet zone control in a reverberant space are now at the fundamental stage; however, these experimental results suggest that the method has great potential.

2.2 Active Minimization of Power in a Reverberant Space

The sound power output from a source can also be reduced in a closed space by using secondary sources. The total power response from all the sources can be made zero by using N secondary sources in the room where N wave modes are excited[2]. In general, however, the number of modes is unknown, and in practical situations the number of secondary sources is smaller than the number of effective modes excited in the room. Thus, generally, the total power response can not reduced to zero, but only to a finite minimum value. This method of reducing the sound power is called "active power minimization" [3][10][11].

The power reduction obtained by a secondary source can be roughly estimated as [10]

$$R = 1 - (1 + M)^{-2} \qquad (2)$$

where M denotes a modal overlap property [12] in the sound field where both the primary and the secondary sources are located. The modal overlap increases, as the frequency becomes high or as the reverberation time becomes short. Therefore, a large reduction in power level is only possible in the low frequency range.

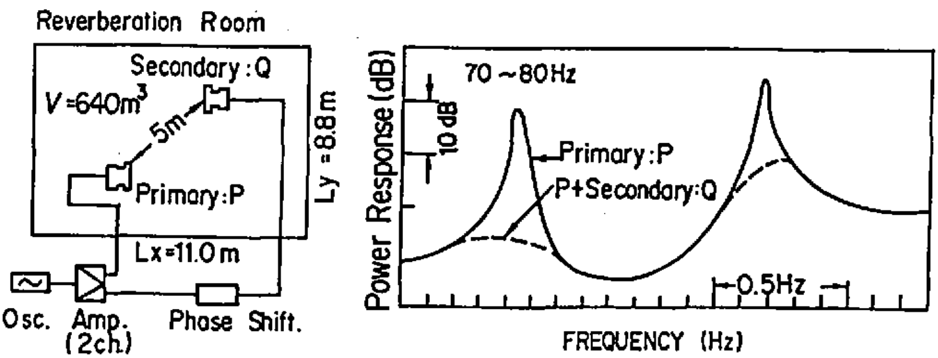


Fig.4 Power response from primary source P and secondary source Q in a large reverberation room

Figure 4 shows the power response of pure tone sources in our large reverberation room. The resonance peaks are greatly reduced as a result of using a secondary source. The formula (2) predicts 8.5 dB of power reduction for our experimental conditions. The agreement with the experimental results is still under study; however, Elliott [13] recently proposed a more effective formula under the very low modal overlap condition.

3. ECHO CANCELLER FOR TELE-CONFERENCING

A tele-conferencing system needs acoustic feedback compensation and room echo suppression for hands-free two-way conversation. A schematic diagram of a tele-conferencing system with an echo canceller is shown in Fig.5. The acoustic echo canceller simulates the TF between a microphone and a loudspeaker in the conferencing room, and cancels only the feed-back signal by using the signal estimated from the echo-path filter (EPF). A finite impulse response (FIR) filter is commonly used for the EPF [14]. The number of taps that the FIR filter should have depends on the reverberation time of the conferencing room.

The frequency characteristics of the TFs, however, are stochastic under the high modal overlap reverberant conditions[12][15][16]. In particular, the arrangements of participants in a tele-conference frequently changes during the conference. Thus, the sound field surrounding the echo-canceller is also time variant. Therefore, the EPF must adapt to such the variations during the conference to cancel the echoes which change irregularly.

Even in such a complex acoustic system, we can expect a predictable averaged behavior in the TFs [17]. The adaptation process may be speed up, if we can use an invariant property. As shown in Figure 6, Makino noticed that the variation in reverberant decay curves also decays exponentially with the same damping coefficient as that of the averaged reverberant decay curve [18]. Consequently, Makino proposed an effective new adaptation algorithm for the EPF based on this random sound field property[19]. Figure 7 clearly demonstrates that this new method greatly reduces the adaptation time needed by the echo canceller compared to conventional ones.

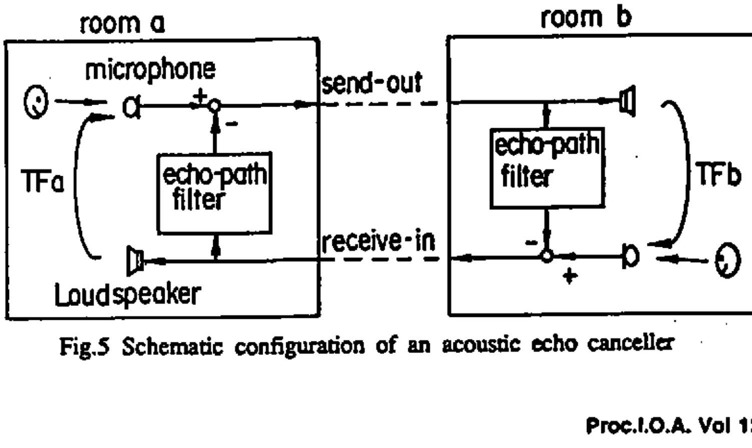


Fig.5 Schematic configuration of an acoustic echo canceller

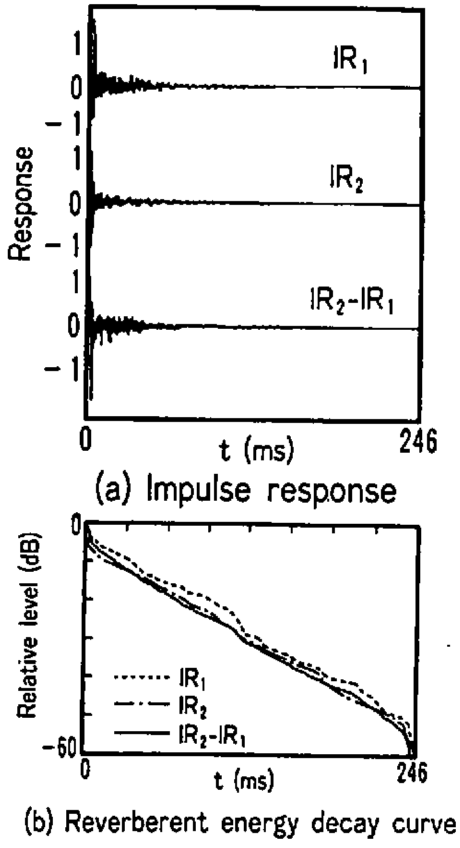


Fig.6 Variation in impulse responses in a reverberant room and reverberation decay curves at different microphone locations

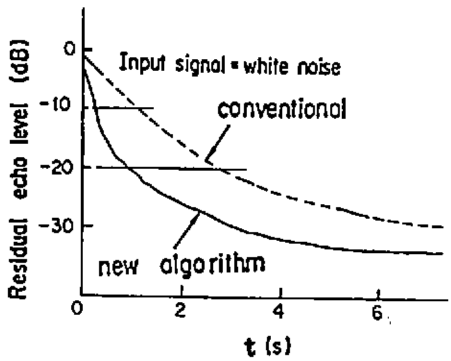


Fig.7 Experimental results for convergence (adaptation) performance of two echo cancellers

4. TRANSFER FUNCTIONS IN REVERBERANT SOUND FIELDS

The design of inverse filters for sound field control and echo-path filters depends on the transfer functions in the sound field. The adaptation process must be assisted by predicting variations in the TFs. Thus, the new developments in the sound field control are exploring a new research area on the prediction of the TF-amplitude and phase properties in random sound fields.

Lyon [17][20][21] investigated the phase characteristics of TFs on machinery diagnostics, and suggested that the averaged phase behavior was predictable according to the arrangement of poles and zeros in TFs, even if the TFs are varied in a random manner. Thus, analysis in poles and zeros of the TFs is important in random sound field control [5][22-24].

4.1. Poles and Zeros of a TF [5][17][20-23]

A TF between two points in a multi-degree-of-freedom system can be expressed as a ratio of polynomials, which are factored according to their roots:

$$H(\omega) = \frac{(\omega - \omega_a)(\omega - \omega_b)(\cdots)}{(\omega - \omega_1)(\omega - \omega_2)(\cdots)} \tag{3}$$

where ω_a, ω_b, ---- are the zeros of H and ω_1, ω_2,--- are its poles. Since the inverse Fourie transform of $H(\omega)$ must be the impulse response of the system, then the imaginary part of the poles must be positive (assuming $e^{i\omega t}$ time-dependency), as shown in Figure 8, but no such restriction applies in general to zeros.

The zeros are distributed symmetrically with respect to the pole-line as shown in Figure 8, assuming the poles have real residues [5][22][24]. Although the pole occurrences on the pole-line are quite irregular [12], and also the symmetrical locations of zeros in the complex-frequency-plane depend randomly on both the source and observation points, the averaged accumulated phase of a TF in a slightly damped system is predictable [17][20-21]:

$$\phi = \frac{\pi}{2} N_p(\omega) \tag{4}$$

where $Np(\omega)$ denotes the expected number of poles up to the angular frequency ω. This accumulated phase is also being studied now under various modal overlap conditions [25].

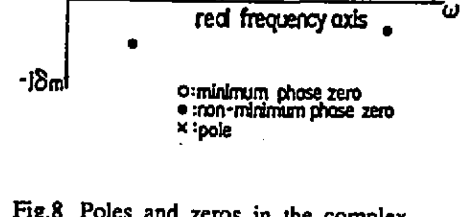


Fig.8 Poles and zeros in the complex-frequency plane of a transfer function

4.2. Pole-zero Modelling of a TF [26]

An FIR is commonly used for an echo-path filter[14]. The number of taps of an FIR filter, however, must become very large, as the reverberation time increases[23]. Therefore, more effective filter design is desirable for an audio echo canceller, even if the digital processors improve rapidly in the near future.

The TFs are characterized by poles and zeros as shown by Equation (3). Thus, it seems reasonable in an echo canceller to design a filter based on pole-zero modelling for the TF identification. The order of such a pole-zero filter depends on the total number of poles and zeros; the number of zeros is roughly on the same order as the number of poles [17][20-21].

Figure 9 compares between the orders of FIR filters and pole-zero filters under a slightly damped condition. We can see a large reduction in the order in the very low frequency region where only a few modes are excited. These results suggest that pole-zero modelling is not suitable for order reduction in echo-path filters; however, the number of effective poles and zeros may be decreased under the high modal overlap reverberant conditions [25]. The results in Figure 9 are still immature, and further investigations are necessary on transfer functions in a reverberant space.

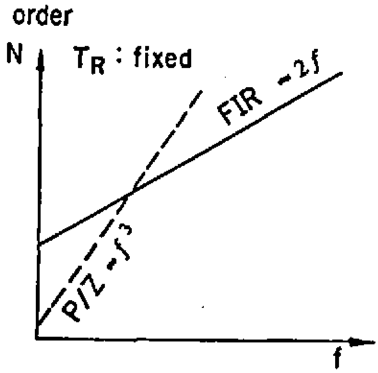


Fig.9 Trends in the orders of Pole/zero and FIR filters

5.CONCLUSION

Recent developments in sound field control are largely due to signal processing technologies; however, a new approach to room acoustics is required now from the point of view of computer-based acoustic control. The fundamental studies described in this article are closely related to the statistical properties and prediction of random sound fields. Much deeper investigation into reverberant fields is necessary in order to create a better acoustic environment.

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