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CANCELLATION OF SIGNAL PICKED UP IN A ROOM BY ESTIMATED SIGNAL

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Introduction

It is required for the oral telecommunication in noisy environment to decrease noise picked up by the microphone. This paper describes the method for decreasing the noise keeping voice signal unchanged and shows the result of experiment successfully carried out. The signal for the cancellation is computed from the signal of noise source. We have realized a real time system for the cancellation at the electical terminal of the microphone. In the system, the convolution between the signal at the noise source and the impulse response from the source to the microphone terminal is subtracted from the output signal of the microphone. Attenuation more than 20 dB is achieved when the room is kept unchanged from the time of measurement of the impulse response. It is easy to obtain 6 dB of attenuation even if someone moves around the microphone.

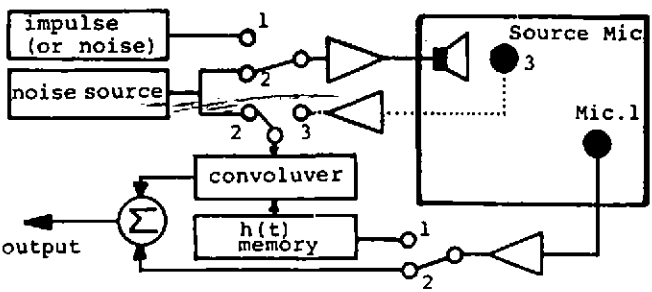

Fig. 1 Sound reinforcement system equipped with signal cancellation system

Principle

Figure 1 shows the schematic diagram of the experimental system equipped with the cancellation system. Before the use of the system, the impulse response h(t) of the transmission path from the driving point of the loudspeaker or a point in front of the noise source to the output terminal of the microphone is measured. When the system is used, the convolution v(t) between the signal x(t) at the source and the impulse response h(t) is computed by high speed digital convoluver as

$$v(t) = x(t)*h(t), \qquad (1)$$

v(t) is subtracted from the output of the microphone y(t) which is the sum of two signals, one of which is the noise and the other is the voice signal s(t) to be reinforced. The remaining signal after the subtraction is in proportion to s(t) if the computation of v(t) is exact.

Preliminary Experiments

Computer simulations and preliminary experiments were carried out before the design of the real time cancellation system. The effect of the duration of the impulse response used for computing convolution on the cancellation is first investigated. The following index is defined for the evaluation of the cancellation effect.

$$P = 10 \cdot \log_{10}[\overline{|y(t)|^2} / \overline{|y(t)-v(t)|^2}] \qquad (2)$$

That is, the index P denotes the ratio of the power of the signal picked up by the microphone to that of the cancelled signal.

First investigated is the effect of the length of impulse response

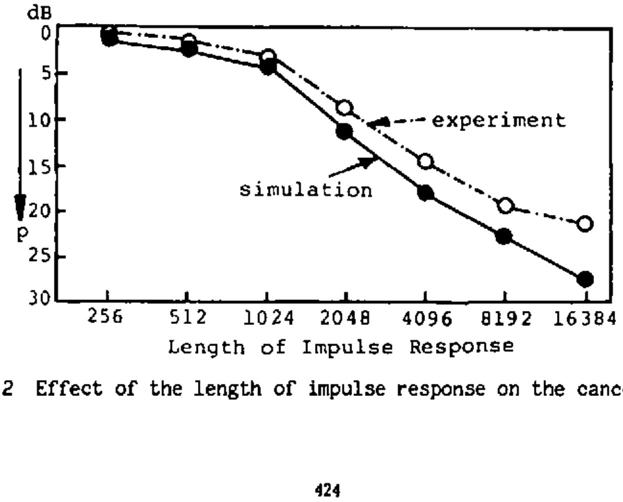

Fig. 2 Effect of the length of impulse response on the cancellation

used for computing the convolution on the cancellation. Figure 2 shows the changes in the index P produced by variations in the length of the impulse response. The magnitude of reverberation of the room decreases by more than 20 dB in 0.52 sec. (4,096 samples). It is found that the experimental results agree well with those of computer simulation and that cancellation more than 20 dB is achieved. Figure 3 shows an example of the spectrum of signal picked up by the microphone and that of the cancelled signal, where the loudspeaker is driven by white noise. It is to be understood that the results of Fig. 2 are obtained after keeping the conditions inside the room unchanged throughout the experiment, but of course, there will be things shifting and people moving around in an actual working room.

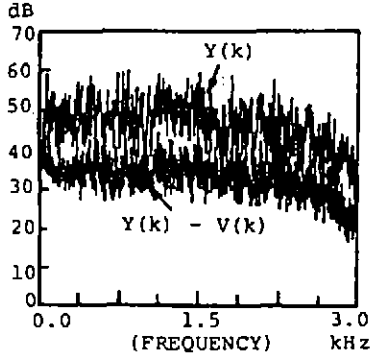

Fig. 3 Power spectrum of the signal picked up by the microphone and that of the cancelled signal

Cancellation System

For the measurement of the impulse response, a train of rectangular pulse is fed to the driving point of the loudspeaker (input terminal of power amplifier) and the signal picked up by the microphone is amplified and is digitized. Every sample of the picked up signal is accumulated truing up the initial point of the response, where the sampling point should be positioned at the center of the rectangular pulse. In our system, the sampling period is 128 micro second, the cut off frequency of the low pass filters used for AD and DA conversion is 2.8 kHz and the length of the impulse response to be used for computation is 0.131 sec. (1022 points).

For the cancellation, the convolution between the signal at the driving point of loudspeaker and the impulse response is computed and is subtracted from the output signal of the microphone. Every sample of the convolution is computed within 128 micro second, that is, 1022 times of multiplication and addition are carried out every 128 micro seconds. Two sets of high speed multiplying accumulators are used. The system is controlled by a micro computer.

Performance of the System

Cancellation of more than 20 dB is achieved in a room where the reverberation time is less than 0.3 second. The index P will decrease for the room having a longer reverberation time. When someone enters the room and/or the door is opened after measuring the impulse response, the value P decreases largely. It is easy, however, to maintain the value more than 6 dB even if some persons move around the microphone and a man approaches up to 30 cm from the microphone so long as the position of the microphone is unchanged. Figure 4 shows an example of experiments relating the number of men with the decrease in the cancellation effect.

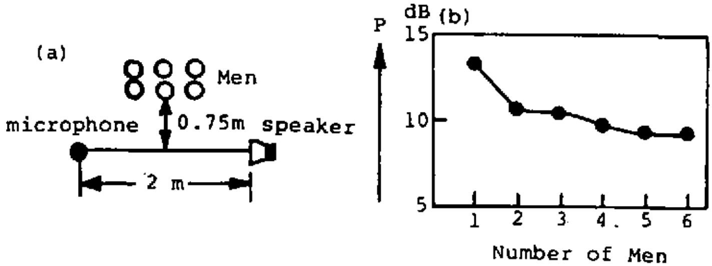

Fig. 4 Relation between number of men and cancellation effect

When P decreases due to a small displacement of the microphone, the value P can be somehow recovered by shifting the impulse response along the time axis, though big compensations can not be expected.

The cancellation system is also useful for preventing the generation of howling in a sound reinforcement system. The increase of more than 6 dB in the amplifier gain is stably obtained by using the cancellation system even if a man moves around the microphone.

Conclusion

The principle, the outline of the experimental system and the experimental results of a signal cancellation system are presented. With this system, the signal picked up in a room is cancelled by the estimated signal. The system can be applied to pitk up voice in a noisy environment. For this purpose, the source signal picked up by Mic. 3 in Fig. 3 should be used for the computation of convolution. The experimental set up is now carried out in which the impulse response should be measured using cross spectra between the signals picked up by Mic. 1 and Mic. 3. The cancellation system is also useful for preventing the generation of howling in a sound reinforcement system.