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ACOUSTICAL LETTER

A study of microphone system for hands-free teleconferencing units

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1. Introduction

A hands-free teleconferencing unit (containing a loudspeaker and directional microphones) is usually used for small teleconferencing meetings.[1,2] An example of the block diagram of such a unit with an acoustic echo canceller (AEC) and four directional microphones is shown in Fig. 1. The acoustic coupling level (between x and e, representing received signal and a transmitted signal, respectively) is too high because the loudspeaker and microphones are close together. To reduce the load placed on the AEC, the acoustic coupling level should be reduced before reaching the AEC.

2. Conventional method

To enable a talker's voice signal to be picked up from a variety of directions, a non-directional microphone or multiple directional microphones are required. The latter approach is usually used to reduce the acoustic coupling level.

In Fig. 1, there is a directional microphone at each corner of the unit. The signal transmitted to the far end talker, e, is obtained by

$$e=y_1+y_2+y_3+y_4. \qquad (1)$$

By positioning the microphones so that their least sensitive direction is oriented toward the loudspeaker, we can minimize the acoustic coupling level. However, actual microphones will still have some sensitivity in the loudspeaker direction, and the acoustic coupling level is increased by a factor of four (*i.e.*, the number of microphones), so it cannot be ignored.

The acoustic coupling level can be reduced by mixing the anti-phase signals. As a result, transmitted signal e is obtained by

$$e=y_1-y_2+y_3-y_4. \qquad (2)$$

In the configuration shown in Fig. 1, the talker is equally distant from microphones 1 and 2. In this situation, the talker's voice is received by two microphones on the same side (microphone 1 and 2) with the same level of power and the same phase, so it is cancelled by mixing the signals as in Eq.(2). As a result, the signal does not reach the far-end talker.

3. Proposed method

To overcome the cancelling problem with the conventional unit, we introduced a 90-degree phase shifter. (A block diagram of a teleconferencing unit using our proposed method with four unidirectional microphones is shown in Fig. 2.) A phase shifter shifts the phase without affecting its power spectrum. The transmitted signal e is now obtained by

$$e=f_0[y_1]+f_{90}[y_2]+f_{180}[y_3]+f_{270}[y_4], \qquad (3)$$

where f_k represents an all-pass filter operation that shifts the phase of the signal by k degrees.

According to Eq. (2) [which represents the conventional method], the transmitted signal is cancelled by mixing two signals with completely inverted phases. With our proposed method, the phase difference between one microphone and the next is 90 degrees, so even if the signal is received by two microphones on the same side with the same level of power and the same phase, the transmitted signal is not cancelled.

4. Simulation testing

We evaluated our proposed method through computer simulation. The distance between the center of the loudspeaker and each unidirectional microphone, r, was 10 cm and the distance between the center of the loudspeaker and the talker, d, was 50 cm (Fig. 3). The talker was at an angle θ of 0, 15, 30, or 45 degrees relative to the center line of the unit.

The frequency response for the transmitted signals with the conventional and proposed units are shown in Figs. 4 and 5, respectively. With the conventional unit, there was no output power at 0 degrees at any frequency. For the other three angles, the output power was very low at low frequencies. With the proposed unit, the power at low frequencies, which is important for speech, was almost flat and did not depend on the direction of the talker.

Figure 6 shows the directional patterns of both units calculated with a pseudospeech signal. The relative positions of the talker, loudspeaker, and unidirectional microphones were the same as for the Fig. 3 data and r and d were again 10 cm and 50 cm, respectively. With the conventional unit, there was no output from the 0, 90, 180, or 270-degree directions. With the proposed unit, the output power was almost flat and did not depend on the direction of the talker.

5. Actual measurement

We next measured the frequency response for trans-

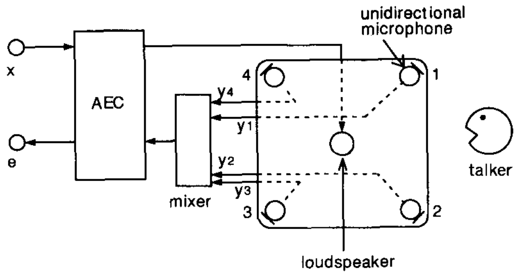

Fig. 1 Block diagram of conventional teleconferencing unit.

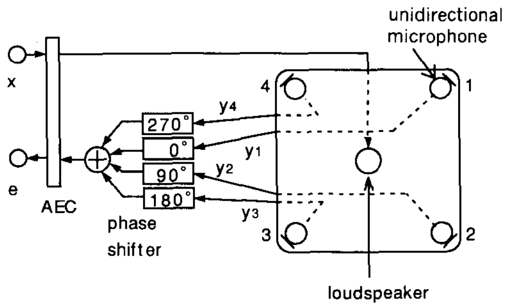

Fig. 2 Block diagram of proposed teleconferencing unit.

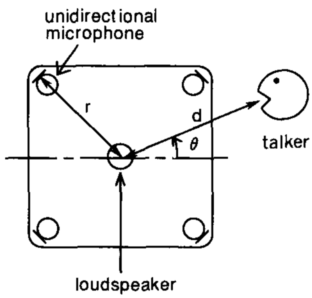

Fig. 3 Conditions for computer simulation.

mitted signals using an actual unit. The frequency bandwidth was set to that of a telephone (from 300 Hz to 3.4 kHz). The talker was set at 0 degrees. The distance between the center of the loudspeaker and each unidirectional microphone was 14.5 cm and the distance between the center of the loudspeaker and the talker was 50 cm. These unidirectional microphones have about 15 dB sensitivity difference on average between maximum and minimum sensitivity. As shown in Fig. 7, the power for low-frequency response with the conventional unit was very low and then it was not flat. The frequency response for the proposed unit was almost flat.

6. Summary

By adding a microphone system with a 90-degree

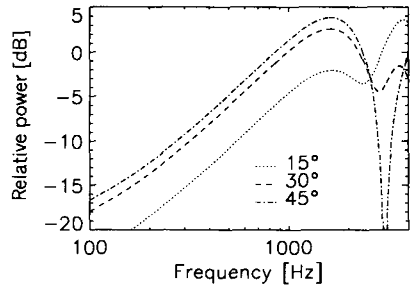

Fig. 4 Simulated frequency response for transmitted signals with conventional method.

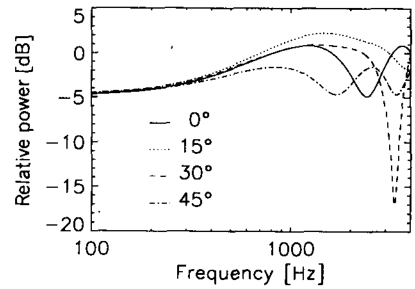

Fig. 5 Simulated frequency response for transmitted signals with proposed method.

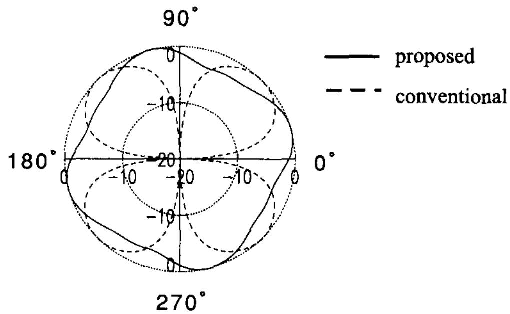

Fig. 6 Directional patterns.

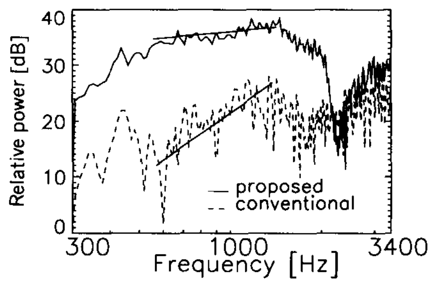

Fig. 7 Measured frequency response for transmitted signals with actual units.

phase shifter to a hands-free teleconferencing unit, we have improved the frequency response for transmitted signal and reduced the acoustic coupling level. Computer simulation and measurement using actual units showed that the output power was flatter with the proposed unit.

References

1) K. Hayashi, A. Hasegawa, T. Sasada, Y. Takaku, and S. Yasunaga, "Development of acoustic teleconferencing equipment using echo cancellation technique," Proc. Spring Meet. IEICE Soc. Jpn., B-962, 408, Mar. (1995) (in Japanese).
2) Y. Haneda, S. Shimauchi, and Y. Kaneda, "A study on hands-free unit with a sub-loudspeaker for acoustic echo reduction," Proc. Spring Meet. IEICE Soc. Jpn., A-10-2, 134, Sep. (1998) (in Japanese).

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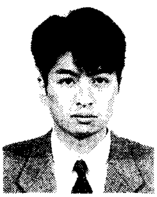

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