A study of multi-microphone system for hands-free teleconferencing units

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ABSTRACT

Hands-free units are used for teleconferences. The units are required to pick up the participants' voices with low degradation and to cancel echoes and howling. This paper presents a multi-microphone system for teleconferencing units using an analogue phase shifter to improve the transmitted signal quality and reduce the acoustic coupling. The results of computer simulation and measurements on a prototype unit showed that our unit reduced the acoustic coupling and flattened the frequency responses of the transmitted signals.

1. INTRODUCTION

A hands-free teleconferencing unit (including a loudspeaker and microphones) is usually used for small-scale teleconferencing meetings [1][2].

Participants sit around the unit and share it. A nondirectional microphone or a few unidirectional microphones are used to pick up voices, which come from various directions. A few unidirectional microphones are used to reduce the acoustic coupling.

Figure 1 shows a block diagram of such a unit including a loudspeaker and four unidirectional microphones with an acoustic echo canceller (AEC), where x, e, and y_n represent the received signal, transmitted signal, and each microphone output signal respectively. In a hands-free environment, the AEC is used for cancelling echoes and howling to make teleconferences comfortable [3]. The acoustic coupling (ratio of x^2 and y_n^2 in Fig. 1) of the teleconferencing unit is very high because the loudspeaker and microphones are close together even if unidirectional microphones are used. The AEC is hard to control when the acoustic coupling is very high.

The acoustic coupling should be reduced before the echo signal reaches the AEC to make the control of the AEC easier. The mixer for y_n should be studied to satisfy its requirement. An adaptive microphone array for noise reduction (AMNOR) is used to reduce undesired signals such as ambient noise [4]. However, it is hard to control both the AMNOR and the adaptive filter of the AEC.

We propose a new multi-microphone system using phase shifters for teleconferencing units to reduce the acoustic coupling with little degradation of transmitted signal.

We implemented these techniques in teleconferencing units. This paper describes the proposed multi-microphone system and shows the results of computer simulations and actual measurements.

2. CONVENTIONAL UNIT

In conventional hands-free teleconferencing units, the transmitted signal, e_l, is simply the sum of the signals

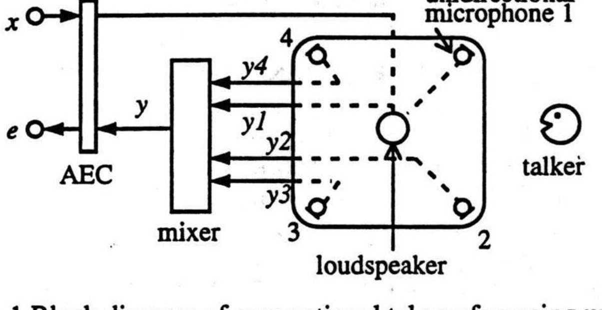

Fig. 1 Block diagram of conventional teleconferencing unit.

from each microphone output, y_n:

$$e_1 = y_1 + y_2 + y_3 + y_4. \quad (1)$$

Actual unidirectional microphones are still a bit sensitive in their least-sensitive direction. Even if their least-sensitive direction is oriented toward the loudspeaker, the acoustic coupling increases in proportion to the number of microphones. It causes a problem because the control of the AEC becomes harder.

In the following, we assume that each microphone that picks up the received signal coming from the loudspeaker with the same power and the same phase. The acoustic coupling can be reduced by using anti-phase signals of two microphone outputs. The transmitted signal, e_2, is calculated by modifying the way the signals are summed, for example:

$$e_2 = y_1 - y_2 + y_3 - y_4. \quad (2)$$

The acoustic coupling can be reduced when each microphone picks up the same signal. But in the configuration shown in Fig. 1, the talker is equally distant from microphones 1 and 2, so the talker's voice is picked up by these two microphones with the same power and same phase as signals y_1 and y_2, respectively. If these two signals, y_1 and y_2, are summed using Eq. (2), they will be cancelled, and the talker's voice will not reach the far-end loudspeaker.

3. PROPOSED UNIT

In Eq. (2) (which we call the conventional method), it is a problem that the transmitted signal will not reach

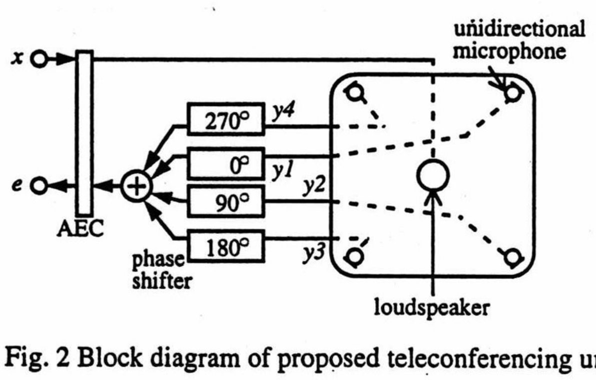

Fig. 2 Block diagram of proposed teleconferencing unit.

the far-end loudspeaker when the talker's voice is picked up by two adjacent microphones with the same power and the same phase. An anti-phased relationship is effective to reduce the acoustic coupling. In the case of Eq. (2), the phase difference between two adjacent microphones is set anti-phased. We studied a multi-microphone system that satisfies the following conditions. The phase difference between two nonadjacent microphones is set anti-phased. When the number of microphones is four and they are set on the circumference of the loudspeaker, there is only one microphone that is not located next to a given one. Therefore, the phase difference between two nonadjacent microphones can be set anti-phased.

Figure 2 shows the block diagram of the proposed multi-microphone system that satisfies the above conditions. We replaced the conventional mixer with one that has phase shifters. The phase of the microphone output signal can be shifted because the quality of the speech is quite insensitive to the phase [5]. The directional pattern of this multi-microphone system has symmetry. There are two sets of anti-phased relationships; each set reduces the acoustic coupling.

The phase shifter shifts the phase of each output signal, y_n, without affecting its power spectrum. The transmitted signal, e_3, is thus

$$e_3 = y_1 * f_0 + y_2 * f_{90} + y_3 * f_{180} + y_4 * f_{270}, \quad (3)$$

where f_k is an all-pass filter that shifts the phase of the signal by k degrees and * indicates convolution.

With the conventional unit, the acoustic coupling can be reduced, but the transmitted signal is cancelled when the talker's voice is picked up by two adjacent microphones on the same side with the same power and the same phase.

With the proposed unit, because the phase difference between two adjacent microphone output signals is 90 degrees, even if the talker's voice is picked up by two adjacent microphones on the same side with the same power and the same phase, the talker's voice is not cancelled. And the acoustic coupling between the received and transmitted signal is reduced by using the proposed multi-microphone system because the echo signal is picked up by all microphones with the same power and the same phase and they are reduced between two non-adjacent microphones.

4. SIMULATION TESTING

We first evaluated the proposed unit by computer simulation. Figure 3 shows the relationship between loudspeaker, microphones, and talker. The distance between the center of the loudspeaker and each microphone, *r*, was 10 cm, and the distance between the center of the loudspeaker and the talker, *d*, was 50 cm. The talker was placed at 0, 15, 30, and 45 degrees relative to the center line of the unit.

Figure 4 shows the frequency responses of the transmitted signals with the conventional unit when pseudo speech was used as the talker's voice. The power at low frequency, which is important for speech, was very low. When the talker was placed at 0 degrees, the talker's voice could not reach the far-end loudspeaker (that frequency response does not appear in Fig. 4).

Figure 5 shows the frequency responses of the transmitted signals with the proposed unit. Each power at low frequency was high and each frequency response was almost flat and independent of the talker's location.

Figure 6 shows the directional patterns of both the conventional and proposed units using pseudo speech as the talker's voice. The relative positions of the loudspeaker, microphones, and talker were the same as before. With the conventional unit, there were insensitive directions at 0, 90, 180, and 270 degrees. With the proposed unit, the directional pattern was almost flat and independent of the talker's position.

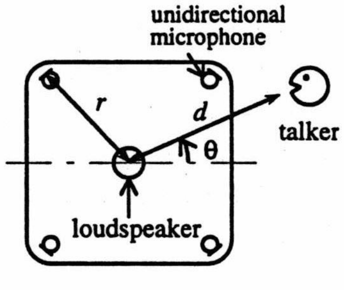

Fig. 3 Conditions for computer simulation.

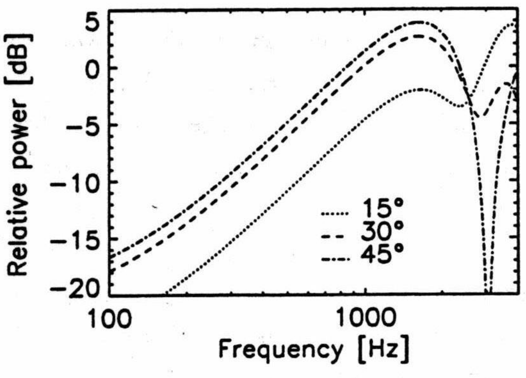

Fig. 4 Simulated frequency responses of transmitted signals with conventional unit.

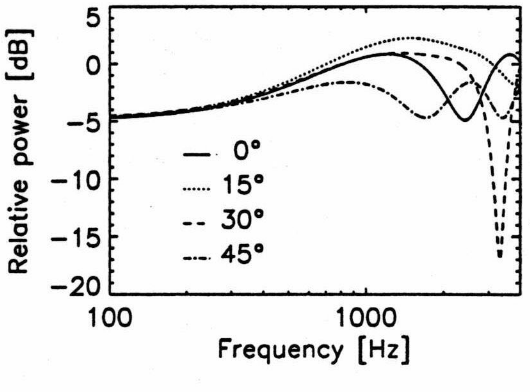

Fig. 5 Simulated frequency responses of transmitted signals with proposed unit.

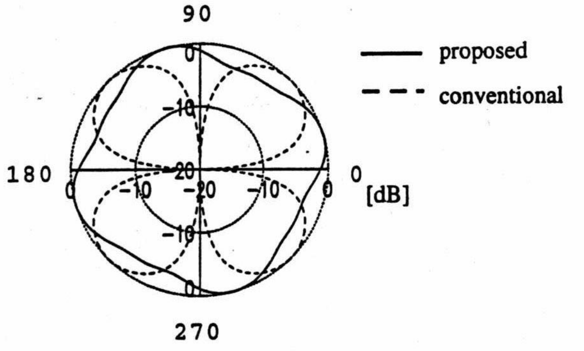

Fig. 6 Directional patterns.

5. IMPLEMENTATION AND ACTUAL MEASUREMENT

5.1. Implementation

We implemented the proposed multi-microphone system in teleconferencing units. The distance between the center of the loudspeaker and each microphone is 14.5 cm. The frequency bandwidth of the unit is 300 through 3400 Hz.

We used an analogue phase shifter, which reduces the acoustic coupling in front of the A/D converter and means that only one A/D converter is needed.

5.2. Actual measurement

We evaluated the speech quality of the transmitted signal using an actual unit. The talker was positioned at 0 degrees relative to the center line of the loudspeaker. The distance between the center of the loudspeaker and the talker was 50 cm.

Figure 7 shows the frequency responses of both the conventional and proposed units. With the conventional unit, the power at low frequency was very low. With the proposed unit, the power at low frequency was high and the frequency response of transmitted signal was almost flat. If the power at low frequency is high, the microphone amplifier can be set low in the teleconferencing unit that handles speech. The acoustic coupling cannot be cancelled completely by the proposed microphone system because the characteristics of actual unidirectional microphones are not all identical. When the microphone amplifiers of both the conventional and proposed units are set the same, the acoustic coupling is theoretically the same. However, the acoustic coupling of the proposed unit is lower than that of conventional one because the microphone amplifier of the proposed unit can be set lower than that of conventional one.

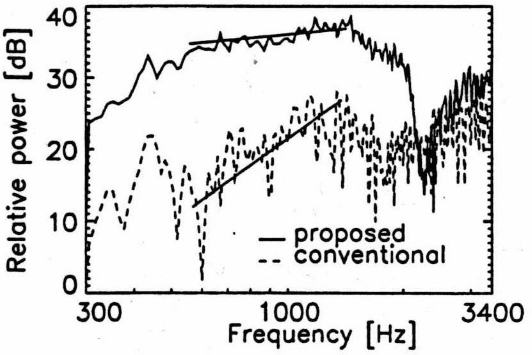

Fig. 7 Measured frequency responses of transmitted signal with actual unit.

6. SUMMARY

We have proposed a multi-microphone system for teleconferencing units and implemented it in a prototype with the AEC. Its acoustic coupling was reduced. The quality of the transmitted signal is improved: the frequency response was flatter. Computer simulation and measurements using actual units showed the effectiveness of the proposed multi-microphone system using phase shifters.

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