

MICROPHONE ARRAY SYSTEM

BACKGROUND AND PRIOR ART

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1. Field of the Invention

This invention relates to a microphone array system.

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2. Description of the Related Art

Normally a microphone is used to acquire a sound. However, the surrounding environment can largely affect the sound acquisition. Background noises and reverberation of the speaker's voice can degrade the acquisition performance, particularly for a distant-talking. For this reason, a short range microphone designed for use near the mouth of the speaker such as a headset microphone or hand microphone has been employed. However, it is uncomfortable to wear the headset microphone on head, while the hand microphone limits the freedom of the speaker as it occupies his/her hands. In addition, it is also inconvenient for multiple speakers for a conference. There has been a demand for a sound input scheme that can allow more freedom and convenience to the speakers.

A known microphone array system has been studied as a potential candidate for sound input scheme that can solve the conventionally encountered inconvenience described above. The microphone array system consists of a set of a plurality of microphones which are arranged at spatially different positions, where the desired speaker can be localized and tracked, and noises can be reduced by synthetic processing of outputs of these microphones.

30 The algorithms for reducing noises by microphone array include conventional delay and sum beamforming, broadband adaptive beamforming and match-filtering etc. However, the delay and sum beamforming can provide low performance, broadband adaptive

beamforming needs a lot of computations and is sensitive to the microphone mismatch and microphone position errors, while match-filtering needs to know the information of the transfer functions between the sound source and microphones in prior.

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SUMMARY OF THE INVENTION

It is accordingly an object of the present invention to provide a microphone array system to overcome at least a part of said inconveniences. It is another object of the present invention to provide a microphone array system to be able to achieve a good spatial selectivity and provide a high signal-to-noise ratio.

This object is solved by providing a first and a second set of microphones for receiving first and second sets of acoustical signals, an adaptive signal processor, a beamforming circuitry and a steering means. Said adaptive signal processor receives a first set of digital signals resulting from said first set of acoustical signals and outputs both, a localization signal and a processed signal. Said beamforming circuitry receives said localization signal and a second set of digital signals resulting from said second set of acoustical signals and outputs a result signal. Said steering means is controlled by said localization signal and adjusts said second set of microphones into a direction where said first set of acoustical signals comes from.

More specifically, according to an embodiment of the present invention, there is provided a steering microphone array system which comprises a first array composed of omnidirectional microphones and a second array composed of directional microphones, preamplifiers, A/D converters, a D/A converter, a video camera (optionally), a small motor to drive the rotation of the fixed directional microphone array, an adaptive signal processor and beamforming circuitry. Said second array composed of directional microphones is of a fixed beam to its direct look direction or such and it is steered (for example: by rotating) by a steering means like a motor and braces according to the localization obtained from said first array composed of omni-directional microphones.

Other objects, features and advantages according to the present invention will be presented in the following detailed description of the illustrated embodiments when read in conjunction with the accompanying drawings.

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BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 illustrates a block diagram of the microphone array system of the present invention;

FIG. 2 illustrates a block diagram of said first array composed of omni-directional microphones;

FIG. 3 illustrates a block diagram of said second array composed of directional microphones;

FIG. 4 illustrates a structure diagram of the arrangement of a j^{th} directional microphone and said motor;

FIG. 5 illustrates a structure diagram of another arrangement for said second array.

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DETAILED DESCRIPTION OF THE INVENTION

FIG. 1 illustrates the block diagram of the microphone array system of the present invention. The system consists of two arrays 1, 2 of microphones. Said first array 1 is composed of m omni-directional microphones OM01, ..., OM0m and said second array 2 is composed of n directional microphones DM01, ..., DM0n.

A first set of acoustical signals received by said m omni-directional microphones OM01, ..., OM0m is sent to a first set of signal converting means comprising a first set of m preamplifiers OP01, ..., OP0m and a first set of m A/D converters OA01, ..., OA0m and converting said (pre-amplified) first set of acoustical signals to a first set of digital signals $x_1(i)$, ..., $x_m(i)$. An adaptive signal processor 4 receives said first set of digital signals $x_1(i)$, ..., $x_m(i)$ and outputs both, a localization signal $od(i)$ and a processed signal $oz(i)$ at first and second output lines 6 and 7, respectively. Said adaptive signal processor 4

determines, whether a voice signal exists, and estimates in a case, where such an acoustical signal exists, from which direction said acoustical signal comes.

5 A second set of acoustical signals (most of these signals originate from the same source {speaker} as said first set of acoustical signals) received by said n directional microphones DM01, ..., DM0n is sent to a second set of signal converting means comprising a second set of n preamplifiers DP01, ..., DP0n and a second set of n A/D converters DA01, ..., DA0n converting said (preamplified) second set of acoustical signals to a second set of digital signals $y1(i)$, ..., $yn(i)$.

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A beamforming circuitry 5 receives said second set of digital signals $y1(i)$, ..., $yn(i)$ and said processed signal $oz(i)$ from said second output line 7, and outputs a result signal $zz(i)$ at an output 8. Said result signal $zz(i)$ is then sent to an output means 11 (for instance: to a connector) through a D/A converter 10.

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A motor 3 (being part of a steering means 2a, 3) is connected to said second array 2, for instance via a brace 2a (said brace 2a also being part of said steering means 2a, 3). Said motor 3 is used to steer a rotation of said second array 2 controlled by said localization signal $od(i)$ from said first output line 6. Said localization signal $od(i)$ from said first
20 output line 6 may, additionally, be used to turn a video camera 9 to focus to the desired speaker.

FIG. 2 illustrates a block diagram of said first array 1 composed of said first set of m omni-directional microphones OM01, ..., OM0m. Said m omni-directional microphones
25 OM01, ..., OM0m receive said first set of acoustical signals and convert it to a first set of electronic signals. Said first set of electronic signals is amplified by said first set of m preamplifiers OP01, ..., OP0m and then converted to said first set of digital signals $x1(i)$, ..., $xm(i)$ by said first set of m A/D converters OA01, ..., OA0m.

30 Within said adaptive signal processor 4 an adaptive voice detection circuit 41 receives said first set of digital signals $x1(i)$, ..., $xm(i)$ and detects, whether a (for example:

human's) voice like signal exists or not. The result of said detection is outputted as a detected signal $v(i)$. If said detected signal $v(i)=0$, no voice signal was detected. If said detected signal $v(i)=1$, a voice signal was detected.

5 A time-delay based localization estimation circuit 42 receives said first set of digital signals $x1(i)$, ..., $xm(i)$ and said detected signal $v(i)$, estimates the localization of the desired speaker (i. e.: the direction, where said first set of acoustical signals comes from), creates and outputs said localization signal $od(i)$ at said first output line 6.

10 The operation of said time-delay based localization estimation circuit 42 is described as follows:

1) If said detected signal $v(i)=0$, said time-delay based localization estimation circuit 42 does not actualize its status (i) and it outputs a previous status $od(i-1)$ of said localization signal od .

15 2) If said detected signal $v(i)=1$, said time-delay based localization estimation circuit 42 actualizes its status (i), estimates the localization of the desired speaker (i. e.: the direction or source, where said first set of acoustical signals comes from), based on time delays between different ones of said omni-directional microphones OM01, ..., OM0m and outputs an actual localization signal $od(i)$ having been actualized compared to
20 its previous status.

A delay and sum beamforming circuit 43 receives said first set of digital signals $x1(i)$, ..., $xm(i)$ and said detected signal $v(i)$, processes a delay and sum beamforming operation and outputs said processed signal $oz(i)$ at said second output line 7.

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FIG. 3 illustrates a block diagram of said second array 2 composed of said second set of n directional microphones DM01, ..., DM0n. Said n directional microphones DM01, ..., DM0n receive said second set of acoustical signals and convert it to a second set of electronic signals. Said second set of electronic signals is amplified by said second set of
30 n preamplifiers DP01, ..., DP0n and then converted to said second set of digital signals $y1(i)$, ..., $yn(i)$ by said second set of n A/D converters, DA01 ..., DA0n.

Said processed signal $oz(i)$ from said adaptive signal processor 4, associated with said first array 1 of omni-directional microphones OM01, ..., OM0m, is delayed at a delay circuit 52 and weighted at a weight circuit 53 to generate a weighted signal $ou(i)$. A fixed look-direction beamforming circuit 51 receives said second set of digital signals $y1(i)$, ..., $yn(i)$ and said weighted signal $ou(i)$ and does beamforming at the look direction (i. e.: the direction or source, where said first set of acoustical signals comes from). Said motor 3 steers said second array 2 into the direction of the desired speaker (i. e., where said first set of acoustical signals comes from) by means of, for example, said braces 2a, controlled by said localization signal $od(i)$ from said first output line 6.

Said fixed look-direction beamforming circuit 5 outputs said result signal $zz(i)$. Said result signal $zz(i)$ goes through said D/A converter 10 and then leaves the microphone array system at said output means 11 (which may be a connector) as a final result signal of the microphone array system.

FIG. 4 illustrates a structure diagram of the arrangement of one of said directional microphones, for instance microphone DM0j, furtheron called as " j^{th} directional microphone", and said motor 3. Said j^{th} directional microphone DM0j is of a directivity pattern DM0j_1 which has a main beam and two nulls (in Fig. 4 only one of said nulls is labeled by the word "NULL"). Said motor 3 is installed in the direction of one of said nulls of said j^{th} directional microphone DM0j in order to reduce the influence due to the rotating noise of said motor 3 into said j^{th} directional microphone DM0j. Another way is to put said motor 3 at the null direction of the whole directivity pattern.

FIG. 5 illustrates a structure diagram of another arrangement for said second array 2. In this arrangement, said directional microphones DM01, ..., DM0n are positioned along an inner circuit. This arrangement is more suitable for near field applications such as a small room, car, computer workstation or the like.

What is claimed is:

1. A microphone array system comprising:
 - a first (1) and a second (2) sets of microphones (OM01, ..., OM0m; DM01, ..., DM0n) for receiving first and second sets of acoustical signals;
 - an adaptive signal processor (4) and a beamforming circuitry (5);
 - a steering means (2a, 3);
 - said adaptive signal processor (4) receiving a first set of digital signals ($x1(i)$, ..., $xm(i)$) resulting from said first set of acoustical signals and outputting a localization signal ($od(i)$) and a processed signal ($oz(i)$),
 - said beamforming circuitry (5) receiving said processed signal ($oz(i)$) and a second set of digital signals ($y1(i)$, ..., $ym(i)$) resulting from said second set of acoustical signals and outputting a result signal ($zz(i)$),
 - said steering means (2a, 3) being controlled by said localization signal ($od(i)$),
 - said steering means (2a, 3) steering said second set (2) of microphones (DM01, ..., DM0n) into a direction where said first set of acoustical signals comes from,
 - said first (1) set of microphones (OM01, ..., OM0m) comprises a plurality of omni-directional microphones, and
 - said second (2) set of microphones (DM01, ..., DM0m) comprises a plurality of directional microphones.
2. The microphone array system according to claim 1, characterized in that said first and second sets of digital signals ($x1(i)$, ..., $xm(i)$; $y1(i)$, ..., $yn(i)$) are generated by first and second sets of signal converting means (OP01, ..., OP0m; OA01, ..., OA0m; DP01, ..., DP0n; DA01, ..., DA0n).
3. The microphone array system according to claim 1, characterized in that said sets of signal converting means (OP01, ..., OP0m; OA01, ..., OA0m; DP01, ..., DP0n; DA01, ..., DA0n) comprise first and second sets of preamplifiers (DP01, ..., OP0m; DP01, ..., DPn), respectively.
4. The microphone array system according to claim 2 or 3, characterized in that

said sets of signal converting means (OP01, ..., OP0m; OA01, ..., OA0m; DP01, ..., DP0n; DA01, ..., DA0n) comprise first and second sets of A/D converters (OA01, ..., OA0m; DA01, ..., DA0n), respectively.

- 5 5. The microphone array system according to claim 1, characterized in that
said adaptive signal processor (4) determines, whether a voice signal exists, and
estimates in a case, that such an acoustical signal exists, from which direction said
acoustical signal comes.
- 10 6. The microphone array system according to claim 5, characterized in that
said adaptive signal processor (4) comprises a voice detection circuit (41) for
receiving said first set of digital signals (x1(i), ..., xm(i)) for detecting, if there is
said acoustical signal and for outputting a detected signal (v(i)).
- 15 7. The microphone array system according to any of claims 5 or 6, characterized in
that said adaptive signal processor (4) comprises a time-delay based localization
estimation circuit (42) for receiving said first set of digital signals (x1(i), ...,
xm(i)) and said detected signal (v(i)), for estimating the localization of the source
of said first set of acoustical signals received and for outputting said localization
20 signal (od(i)).
8. The microphone array system according to any of claims 5, 6 or 7, characterized
in that said adaptive signal processor (4) comprises a delay and sum beamforming
circuit (43) for receiving said first set of digital signals (x1(i), ..., xm(i)) and said
25 detected signal (v(i)) and for outputting said processed signal (oz(i)).
9. The microphone array system according to claim 1, characterized in that
said beamforming circuitry (5) is arranged to combine said second set of digital
signals (y1(i), ..., ym(i)) with said processed signal (oz(i)).
- 30 10. The microphone array system according to claim 9, characterized in that

said beamforming circuitry (5) comprises a delay circuit (52) to delay said processed signal (oz(i)) received.

11. The microphone array system according to claim 9 or 10, characterized in that
5 said beamforming circuitry (5) comprises a weight circuit (53) to weight said delayed processed signal (oz(i)) and outputting it as a weighted signal (ou(i)).

12. The microphone array system according to claim 11, characterized in that
10 said beamforming circuitry (5) comprises a fixed look-direction beamforming circuit (51) receiving said weighted signal (ou(i)) and said second set of digital signals (y1(i), ..., ym(i)) for beamforming into said direction, where said first set of acoustical signals comes from.

13. The microphone array system according to any of claims 9 to 12, characterized in
15 that said fixed look-direction beamforming circuit (51) is arranged to output said result signal (zz(i)).

14. The microphone array system according to claim 1, characterized in that
20 said result signal (zz(i)) is input to a D/A converter (10).

15. The microphone array system according to claim 14, characterized in that
an output of said D/A converter (10), which is arranged to carry an analogue
signal corresponding to said result signal (zz(i)), is connected to an output
connector means (11).

16. The microphone array system according to claim 15, characterized in that
25 said output means (11) is a connector.

17. The microphone array system according to claim 1, characterized in that
30 a video camera (9) is controlled by said localization signal (od(i)) for steering it into a direction where said first set of acoustical signals comes from.

18. The microphone array system according to claim 1, characterized in that
said steering means (2a, 3) comprises a motor (3) for processing said steering.
- 5 19. The microphone array system according to claim 1 or 18, characterized in that
said steering means (2a, 3) comprises a brace (2a) engaged with said second set
(2) of microphones (DM01, ..., DM0n).