

An Adaptive Sound Sensing System

--- Intelligent Fault sound detection system ---

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ABSTRACTS

An adaptive sound sensing system incorporated multiple microphone array and real time digital signal processing algorithm is described.

The objectives of the system are to receive the useful sound signal from damaged components or faulty system out of various noise environments with better S/N ratio.

This adaptive sound sensing system is described as an example of one approach to advanced intelligent sensing system.

The results of implementation proves effectiveness in S/N improvement in noisy environment.

1. BACKGROUND

Human sensory system can be considered very advanced example of intelligent sensing system.

We can recognize our names or certain special words which are interested if we hear them even in noisy environments. These characteristics are referred as "Cock tail party effect". "How we can do?" the mechanism was not seemed to be clarified. We can also notice the abnormal sound from components of faulty machines out of operating noise.

The sophisticated mental recognition system may contribute to the function, but physical level of our auditory system can concentrate their power on the interested sound source and improve their S/N ratio.

This study was promoted by our motivation to realize a physical model of "Cock tail party effect".

2. STRUCTURE OF INTELLIGENT SENSING SYSTEMS

Intelligent sensing systems have a hierarchical structure having multi layer as ordinary system. On the uppermost layer the most highly intelligent information processing is done. The function of processing is centralized like human brain. Processed information is rather abstract, and it is not independent on physical structure of sensors. On the contrary, group of intelligent sensors on the lowest layer gather informations from the external objects like our distributed sensory organs. Signal processing of these intelligent sensors is done in distributed and parallel manner. Processed informations are strongly dependent on principles and structures of the sensors.

There are intermediate signal processing functions in the

middle layer. A function of the intermediate signal processing is an integration of signals from multiple intelligent sensors. Another function is tuning of parameters of the sensors to optimize the total system performance.

In general algorithm of the information processing is more flexible in the higher layer and less flexible in the lower layer.

3. ROLE OF INTELLIGENCE IN SENSORS

The sensor intelligence performs distributed signal processing at the lower layer in the hierarchy of the sensing system.

The role of signal processing function of intelligent sensors can be summarized as follows;

- 1) Extraction of essential features from the objects.
- 2) Reinforcement of sensor proper characteristics.

The important objective of the signal processing is to improve the signal selectivity of sensors. This includes above two roles of sensor intelligence.

The extracted features are converted to the recognition of the situation by upper layer intelligence. By the processing in the upper layer, the output signals from multiple sensors are combined or integrated.

4. PRINCIPLES OF INTELLIGENT SOUND SENSING SYSTEM

Our intelligent sound sensing system has two layer structure as shown in Fig.1.

The lower layer of system consists of one set of multiple microphones and two sets of digital signal processing unit connected to microphones. It has a computer in the upper layer. The computer evaluates the system S/N and tunes parameters of the processing units. The properties of the processing units are controlled by a computer. The outputs from the signal processing units are used as the intermediate output of the system. The computer calculates and compares the two intermediate outputs from the system, and tunes the parameters of processing units so as to increase S/N ratio.

The intermediate output from the optimized processing unit is used the final output of the system.

The location of abnormal sound source is not given. Both waveforms of signal and noise sources are unknown. However, some feature of the signal is given as cue to discriminate the signal out of noise. The computer calculate S/N on the basis of the cue.

5. ALGORITHM FOR OPTIMIZATION

Here the signal is assumed nonstationary sound of which power varies time to time. The noise can be stationary sound. Total power of these sounds can be denoted as a linear combination of signal power P_s and noise power P_n as Eq.1.

$$P = a \cdot P_s + b \cdot P_n \quad (1)$$

We measure the average of power during time interval $[t_1 - t_1 + dt]$ & $[t_2 - t_2 + dt]$ obtained by the use of system 1 & 2. Let the average power of $[t_1 - t_1 + dt]$ and $[t_2 - t_2 + dt]$ obtained by system 1 be $P_1(t_1)$ and $P_1(t_2)$ respectively. Let the similar average power sensed by system 2 be $P_2(t_1)$ and $P_2(t_2)$.

Then, we get

$$P_1(t_1) = a_1 * P_s(t_1) + b_1 * P_n \quad (2)$$

$$P_1(t_2) = a_1 * P_s(t_2) + b_1 * P_n \quad (3)$$

$$P_2(t_1) = a_2 * P_s(t_1) + b_2 * P_n \quad (4)$$

$$P_2(t_2) = a_2 * P_s(t_2) + b_2 * P_n \quad (5)$$

From Eq. (2) -- (5), we can calculate,

$$E_1 = \frac{P_1(t_1) - P_1(t_2)}{P_2(t_1) - P_2(t_2)} = \frac{a_1}{a_2} \quad (6)$$

then we can compare the power level of the desired signal component processed in the two systems.

If $E_1 > 1$, system 1 get more signal than system 2.

We can calculate E_2 also from Eq. (2) -- (5), and we can compare S/N of two systems by calculating E_2 .

$$\begin{aligned} E_2 &= \frac{P_1(t_1) + P_1(t_2)}{P_2(t_1) + P_2(t_2)} * \frac{P_2(t_1) - P_2(t_2)}{P_1(t_1) - P_1(t_2)} \\ &= \frac{P_s(t_1) + P_s(t_2) + 2(b_1/a_1)P_n}{P_s(t_1) + P_s(t_2) + 2(b_2/a_2)P_n} \end{aligned} \quad (7)$$

If $E_2 < 1$, then S/N [system 1] $>$ S/N [system 2].

If $E_2 > 1$, then S/N [system 1] $<$ S/N [system 2].

Now we can know which parameters of system should be tuned to improve S/N, and to get more signal.

A hill climb algorithm is used to optimize the processor characteristics in the system 1 or 2. Step and step adjustment of processor characteristics approach to optimum condition in environment. Thus total system adapt their performance to given environments and realize maximum S/N.

The system works as an automatic focusing system in which the focus of multiple microphones coincides with the signal sound source and noises from other sources are canceled out or suppressed.

More sophisticated and rapid algorithm is under development and it may be utilized in the future version.

6. EXPERIMENTAL IMPLEMENTATION

An experimental adaptive sound sensing system hardware is shown in Fig.2. Six microphones are arbitrarily installed in our laboratory and each of them are connected to the A/D converter(16 bits, sampling frequency = 44.1 kHz). The outputs of an A/D converter is processed by two sets of digital filter. The outputs of six digital filters are combined and converted to the intermediate output describing the average power by a personal computer.

The personal computer calculates the ratio $E1$ and/or $E2$, then evaluates the performances of the digital signal processor, and tunes the parameters of digital filter in the processor.

7. EXPERIMENTAL RESULTS

Fundamental function of the system are tested in anechoic room. A point sound source of amplitude modulated white noise is used as the signal source. Unmodulated white noise source and a buzzer are used as stationary noise source.

An example of our test results is shown in Fig.3. Gains of six digital filters are changed and optimized in response to exchange of different noise source. Improvement of S/N is also observed.

8. CONCLUSION

An adaptive sound sensing system is proposed as an aspect of intelligent sensing system.

The position or waveforms of target signal sound source is not given. However feature of the target sound is given to the system as the cue to recognize the abnormal sound signal in noisy environment.

The system optimize its transfer characteristics automatically so as to realize better S/N ratio.

Basic function is confirmed experimentally. In the experiment amplitude modulated white noise is assumed as the target signal and stationary sound is used as its noise.

Now more sophisticated and fast algorithm for signal recognition is tested by computer simulation.

Possible applications of the system are detection of abnormal sound from failed machine or human voice in noisy environment.

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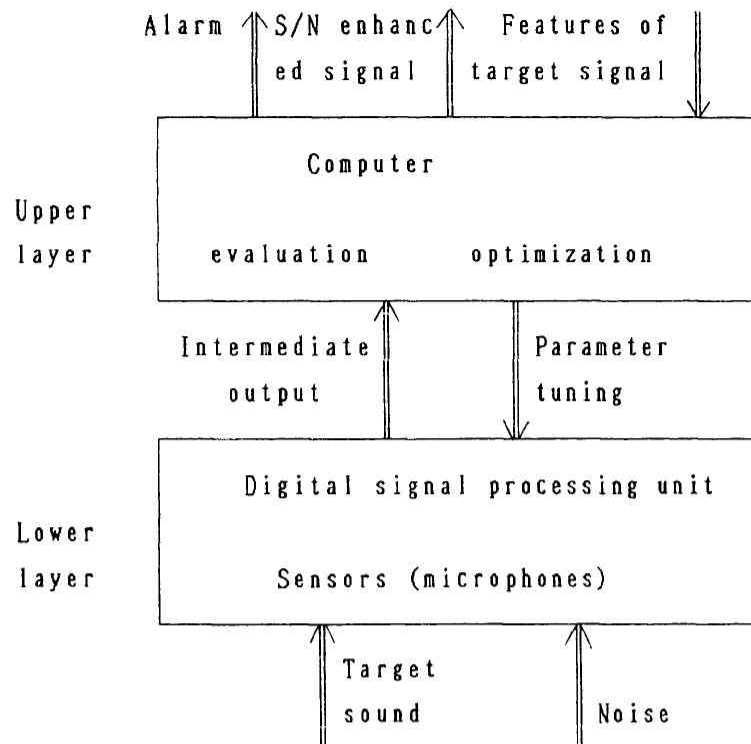


Fig.1 Structure of the intelligent sound sensing system

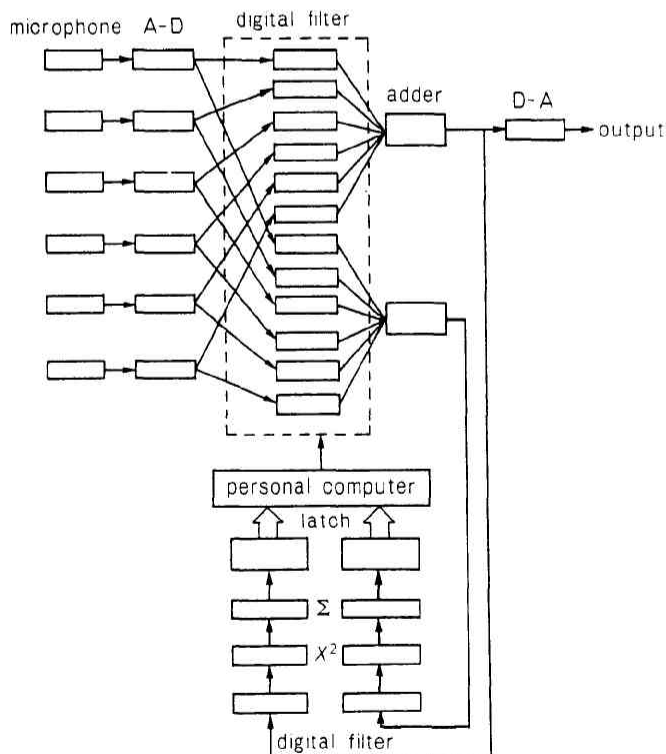


Fig.2 Adaptive sound sensing system hardware

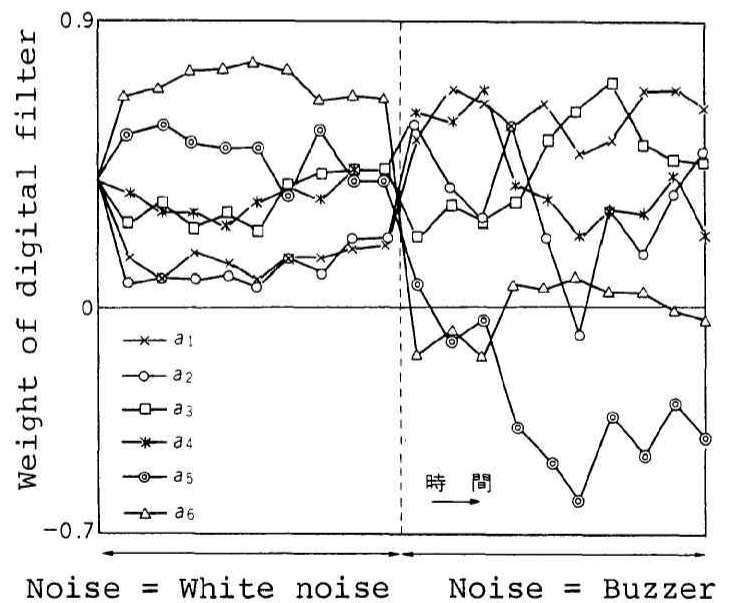


Fig.3 Experimental results