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Emission timing control method for improving signal to interference ratio on public address system

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ABSTRACT

In order to reduce disaster damage, there are several media such as email, TV, radio and a public address system with outdoor loudspeakers to deliver emergency announcements to the people at risk. The Great East Japan Earthquake revealed the importance of a public address system with outdoor loudspeakers for emergency announcements and the difficulty of its intelligibility. To achieve a public address system with sufficient intelligibility for residents, one important factor is to suppress sound overlap caused by reflection from terrain or buildings, or caused by the simultaneous emission of other loudspeakers. Recent Internet technologies and devices will contribute to avoid sound overlap caused by the simultaneous emission of other loudspeakers. In this paper, a method of controlling emission timing to minimize the sound overlap caused by the simultaneous emission of other loudspeakers of a public address system is proposed. This method focused on the transmission of a signal between the input and output of a public address system. Under the assumptions that the attenuation and delay factor between a loudspeaker and a listening point depend only on the distance and the directivity of each node is omnidirection, the optimal emission timing for a pair of nodes is expressed in terms of the signal to interference ratio. Then, on the basis of the above optimal emission timing, the emission timing for a set of nodes in a management area is determined. In the proposed method, each node collects and shares the positions of each node obtained by the Global Positioning System via a computer network to determine the emission timing, and then taking into account whether other nodes emit sound or not each node autonomously determines its emission timing and the delay time for other nodes. The proposed method can be implemented without any dedicated hardware and can, therefore, be applied to existing loudspeakers. The delay and error factors in a public address system employing the proposed method are also modeled and estimated.
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1. Introduction

In order to reduce disaster damage, there are several media such as email, TV, radio [1,2] and a public address system with outdoor loudspeakers shown in Fig. 1 to deliver emergency announcements to the people at risk. In the Great East Japan Earthquake, among residents who could obtain evacuation announcements in the affected prefectures, 45% of them obtained it from outdoor loudspeakers. However, among those who obtained evacuation announcements from outdoor loudspeakers, 56% of them answered they could clearly hear the announcements [3]. Therefore, the Committee for Technical Investigation on Countermeasures for Earthquakes and Tsunamis Based on the Lessons Learned from the "2011 off the Pacific coast of Tohoku Earthquake" suggests improvement of intelligibility of a public address system as one of the countermeasures to mitigate tsunami damage [4]. To achieve addressing with sufficient intelligibility for residents, factors that degrade the intelligibility of radiated sound must be minimized. One important factor is sound overlap at observation points. In particular, sound overlap caused by reflection from terrain or buildings, or caused by the simultaneous emission of other loudspeakers, is known to degrade intelligibility [5–7]. As demonstrated by the hearing ability of humans, and illustrated by the cocktail party effect, intelligibility depends not only on the power ratio between the target signal and interference but also on the context of the speech, the characteristics of the signal, the direction of arrival, and word familiarity. However, in the previous work it was shown to be possible for intelligibility to be estimated from the power ratio of overlapping sounds [8]. In accordance with the result of this previous work, in this paper, sound overlap is defined by the power ratio of the signal arrived from one node to the signals arrived from other nodes. Therefore, in this paper, the impact of the context of the speech,

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Fig. 1. Outdoor loudspeaker of a public address system for emergency use.

the characteristics of the signal, the direction of arrival, and word familiarity are not considered. Because the signals emitted by the nodes of a public address system in one area are the same, when the phase of overlapped sounds is the same, the sound is emphasized rather than masked. However, in the actual situation, this kind of phase synchronization can be considered as a rare case. Therefore, in this paper, the impact of phase is not considered. Based on this definition of sound overlap, sound arriving from adjacent loudspeakers can be considered to degrade intelligibility. Therefore, a method of suppressing sound overlap between adjacent loudspeakers is required.

One simple way is to set loudspeakers so that each service area of them does not have overlap. However, since there are loudspeakers already installed, it costs so much to move them. For example, at the end of 2001, 65% of all the local governments of cities, towns, and villages in Japan was equipped with the broadcast system for disaster [9].

In this paper, a method for controlling the timing of emission to reduce sound overlap in a public address system is proposed. In a public address system with loudspeakers, an input signal is transmitted via a path consisting of two portions. The first portion is the path between the input and output of the public address system. The second portion is the path between a loudspeaker and a listening point. The latter path is affected by many factors such as rainfall, absorption by air and terrain, the distribution of temperature, humidity, reflection by terrain and buildings, and wind. In particular, previous works showed that open-air sound transmission between a loudspeaker in a public address system and a listening point are affected by reflection [6] and wind [10,11]. In this paper, the latter path is simplified such that attenuation and delay factors of the impulse response between a loudspeaker and a listening point only depend on the distance. The proposed method focuses on controlling the emission timing for the former path, the path between the input and output of the public address system. The proposed method delays start time of emission of each node to avoid sound overlap by following processes.

Each node collects and shares the positions of each node obtained by the Global Positioning System (GPS) via a computer network to determine the emission timing. An input signal can be fed into one of each node in the same management area. The input signal is transmitted to other nodes, then each node detects the start time of a sound event from the transmitted signal. After detecting the start of a sound event, a node sends a query to other nodes to obtain their status of emission. On the basis of response from other nodes, each node autonomously determines its emission timing.

This paper is organized as follows. In Section 2, the optimal emission timing for a pair of nodes in terms of the signal to interference ratio is derived. Then a model for determining the emission timing for a set of nodes in a management area is shown. In Section 3, to accomplish the optimal emission timing derived in Section 2, a method of controlling emission timing for a public address system is proposed. In Section 4, the delay and error factors in a public address system employing the proposed method are investigated. In Section 5, the proposed method is implemented on a laptop PC without any dedicated hardware and a demonstration is conducted. Conclusions are given in Section 6.

2. Model for determining emission timing for public address system

2.1. Signal to interference ratio in public address system

The amplitude of an input signal fed into a public address system at time t is denoted by

$$a(t) = \begin{cases} A, & 0 \leqslant t \leqslant N \\ 0, & \textit{otherwise} \end{cases}, \tag{1}$$

where $A \in \mathbb{R}$ and N is the duration of the input signal. The amplitude of the output signal of node $\mathbf{s}_n$ at t, denoted by $b_n(t)$, is obtained as

$$b_n(t) = g_n a(t - \tau_n), \tag{2}$$

where g_n is the gain factor and τ_n is the delay factor of $\mathbf{s}_n$. Then the amplitude of an observed signal at a listening point in the service area of $\mathbf{s}_n$ at time t, denoted by $y_n(t)$, is calculated as

$$y_n(t) = \sum_{k=-\infty}^{\infty} b_n(k) h_n(t-k), \tag{3}$$

$$= \sum_{k=-\infty}^{\infty} g_n a(k - \tau_n) h_n(t-k), \tag{4}$$

where $h_n(t)$ is the impulse response between $\mathbf{s}_n$ and the listening point. $y_n(t)$ can be rewritten as

$$y_n(t) = g_n \alpha_n a(t - \tau_n - \beta_n), \tag{5}$$

where α_n is the attenuation factor and β_n is the delay factor of $h_n(t)$. α_n and β_n depend on many factors such as distance attenuation, rainfall, wind, reflection by terrain and buildings, absorption by air and terrain, the distribution of temperature, and humidity. In this paper, α_n and β_n are assumed to depend only on the distance as follows:

$$y_n(t) = g_n \alpha(r_n) a\left(t - \tau_n - \frac{r_n}{c}\right), \tag{6}$$

where r_n is the distance between $\mathbf{s}_n$ and a listening point and c is the speed of sound. Therefore, the signal to interference ratio at time t regarding the sound from $\mathbf{s}_n$ as the signal and the sound from other nodes $\mathbf{s}_m (m \neq n)$ as interference, denoted by $\mathrm{SIR}_n(t)$, can be obtained as follows:

$$\mathrm{SIR}_n(t) = 20\log_{10} \frac{|y_n(t)|}{\sum_m |y_m(t)|}, \tag{7}$$

$$= 20\log_{10} \frac{g_n \alpha(r_n) |a(t - \tau_n - \frac{r_n}{c})|}{\sum_m g_m \alpha(r_m) |a(t - \tau_m - \frac{r_m}{c})|}. \tag{8}$$

The vector of the optimal delay factor for each node in terms of the signal to interference ratio, $\tau_{\mathbf{opt}}$, is then represented as

$$\boldsymbol{\tau}_{\mathrm{opt}} = \underset{\tau_n, \tau_m}{\operatorname{argmax}} \sum_n \sum_S \sum_{t \in \mathbf{t_n}} \mathrm{SIR}_n(t), \tag{9}$$

where $\mathbf{t_n}$ is the vector of time when $y_n(t) \neq 0$ and $\sum_S$ is the summation over listening points in the service area of $\mathbf{s}_n$.

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