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6. BEA specializes in the research and delivery of materials from the Library of Congress.
7. On September 26, 2025, I was asked by the requester Morrison and Foerster LLP to obtain a scan from the following book: Springer handbook of speech processing (2008).
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Signed:

C. Vassilkova

Springer Handbook of Speech Processing



From common consumer products such as cell phones and MP3 players to more sophisticated projects such as human-machine interfaces and responsive robots, speech technologies are now everywhere. Many think that it is just a matter of time before more applications of the science of speech become inescapable in our daily life. This **Springer Handbook of Speech Processing** is meant to play a fundamental role for sustainable progress in speech research and development. It targets three categories of readers: graduate students, professors and active researchers in academia and research labs, and engineers in industry who need to understand or implement some specific algorithms for their speech-related products. The handbook could also be used as a sourcebook for one or more graduate courses on signal processing for speech and different aspects of speech processing and applications. A quickly accessible source of application-oriented, authoritative and comprehensive information about these technologies, it combines the established knowledge derived from research in such fast evolving disciplines as Signal Processing and Communications, Acoustics, Computer Science, and Linguistics.

Key Topics

- › Production, Perception, and Modeling of Speech
- › Signal Processing for Speech
- › Speech Coding
- › Text-to-Speech Synthesis
- › Speech and Speaker Recognition
- › Language Recognition
- › Speech Enhancement
- › Multichannel Speech Processing

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Benesty
Sondhi
Huang
Editors



Springer
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of
Speech
Processing

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best viewed with Adobe Reader 8



System Requirements

- Windows 95/98/ME 32 MB RAM, Windows NT 4/2000/XP 64 MB RAM, Windows Vista 512 MB RAM
- MAC OS 9 or higher, 64 MB RAM, 333 MHz
- LINUX Pentium I /166 MHz, 64 MB RAM

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Springer Handbook of

Speech Processing



Benesty
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**Springer Handbook
of Speech Processing**

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Springer Handbook of Speech Processing

Jacob Benesty, M. Mohan Sondhi, Yiteng Huang
(Eds.)

With DVD-ROM, 456 Figures and 113 Tables

 Springer

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Foreword

Over the past three decades digital signal processing has emerged as a recognized discipline. Much of the impetus for this advance stems from research in representation, coding, transmission, storage and reproduction of speech and image information. In particular, interest in voice communication has stimulated central contributions to digital filtering and discrete-time spectral transforms.

This dynamic development was built upon the convergence of three then-evolving technologies: (i) sampled-data theory and representation of information signals (which led directly to digital telecommunication that provides signal quality independent of transmission distance); (ii) electronic binary computation (aided in early implementation by pulse-circuit techniques from radar design); and, (iii) invention of solid-state devices for exquisite control of electronic current (transistors – which now, through microelectronic materials, scale to systems of enormous size and complexity). This timely convergence was soon followed by optical fiber methods for broadband information transport.

These advances impact an important aspect of human activity – information exchange. And, over man's existence, speech has played a principal role in human communication. Now, speech is playing an increasing role in human interaction with complex information systems. Automatic services of great variety exploit the comfort of voice exchange, and, in the corporate sector, sophisticated audio/video teleconferencing is reducing the necessity of expensive, time-consuming business travel. In each instance an overarching target is a user environment that captures some of the naturalness and spatial realism of face-to-face communication. Again, speech is a core element, and new understanding from diverse research sectors can be brought to bear.

Editors-in-Chief Benesty, Sondhi and Huang have organized a timely engineering handbook to answer this need. They have assembled a remarkable compendium of current knowledge in speech processing. And, this accumulated understanding can be focused upon enlarging the human capacity to deal with a world ever increasing in complexity. Benesty, Sondhi and Huang are renowned researchers in their own right, and they have attracted an international cadre of over 80 fellow authors and collaborators who constitute a veritable *Who's Who* of world leaders in speech processing research. The resulting book provides under one cover authoritative treatments that commence with the basic physics and psychophysics of speech and hearing, and range through the related topics of computational tools, coding, synthesis, recognition, and signal enhancement, concluding with discussions on capture and projection of sound in enclosures. The book can be expected to become a valuable resource for researchers, engineers and speech scientists throughout the global community. It should equally serve teachers and students in human communication, especially delimiting knowledge frontiers where graduate thesis research may be appropriate.

Warren, New Jersey
October 2007

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50. Microphone Arrays

G. W. Elko, J. Meyer

This chapter introduces various types of microphone array beamforming systems and discusses some of the fundamental theory of their operation, design, implementation, and limitations. It is shown that microphone arrays have the ability to offer directional gains that can significantly improve the quality of signal pickup in reverberant and noisy environments.

Hands-free audio communication is now a major feature in mobile communication systems as well as audio and video conferencing systems. One problem that becomes evident to users of these systems is the decrease in communication quality due to the pickup of room reverberation and background noise. In the past, this problem was dealt with by using microphones placed close to the desired talker or source. Although this simple solution has proven to be quite effective, it also has its drawbacks. First, it is not always possible or desirable to place the microphone very close to the talker's mouth. Second, by placing the microphone close to the talker's mouth, one has to deal with rapid level variation as the talker moves his or her mouth relative to the microphone. Third is the negative impact of speech plosives (air-flow transients generated by plosive sounds) and forth, microphone structure-borne handling noise has a detrimental effect. Finally, for directional microphone elements, there is also a nearfield

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proximity effect (where the frequency response of the microphone is modulated by the relative position of the microphone to the mouth). With these issues in mind, it is of interest to investigate other potential solutions. One solution is to use beamforming microphone arrays, which can offer significant directional gain so as to result in similar audio performance to that of closely placed microphones.

50.1 Microphone Array Beamforming

The propagation of acoustic waves in space is a function of both space and time coordinates. In general, the mathematical representation of a propagating acoustic wave is a four-dimensional function: three spatial dimensions and one time variable. Acoustic wave propagation can be modeled by a linearized scalar acoustic wave equation that describes the relationships between the physical quantities of acoustic pressure and density variation of the medium [50.1],

$$\nabla^2 p = \frac{1}{c^2} \frac{\partial^2 p}{\partial t^2}, \quad (50.1)$$

where p is the instantaneous acoustic pressure fluctuation of the sound and c is the propagation speed of sound in the medium. There are a few major underlying assumptions in the derivation of (50.1), but this simplified linear model is adequate for laying the foundation for array beamforming of acoustic signals. The dimen-

sionalities of the wave equation can easily be seen from (50.1), where p is a function of three space variables and one time variable. Therefore the scalar acoustic pressure field in space can be represented as $p(\mathbf{r}, t)$, where $\mathbf{r}$ is the measurement position in the acoustic field and t is the time dependence. The scalar acoustic pressure field must satisfy (50.1) at all points in space. By using the spatial Fourier transform, the acoustic field can also equivalently be represented in the wavevector–frequency domain. This representation has been widely used in the fields of geophysics, structural, underwater and aeroacoustics, and can be quite useful in the analysis and design of microphone array beamformers.

Applying the four-dimensional Fourier transform to the time–space scalar acoustic field yields,

$$P(\mathbf{k}, \omega) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} p(\mathbf{r}, t) e^{-i(\omega t - \mathbf{k}^T \mathbf{r})} d\mathbf{r} dt, \quad (50.2)$$

where the acoustic pressure P is capitalized to indicate a transform to the frequency domain, $\mathbf{k}$ is the wavevector and the superscript ‘T’ represents the transpose operator. It may seem that this transformation has unnecessarily complicated the description of the scalar acoustic pressure field, but as will be seen later, it has transformed the field description into a form that makes the analysis of acoustic field space–time functions analogous to the field of multidimensional signal processing. Since the Fourier transform is a linear transformation, one can write an inverse relationship,

$$p(\mathbf{r}, t) = \frac{1}{(2\pi)^4} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} P(\mathbf{k}, \omega) e^{i(\omega t - \mathbf{k}^T \mathbf{r})} d\mathbf{k} d\omega, \quad (50.3)$$

Equation (50.3) directly shows that any acoustic field $p(\mathbf{r}, t)$ can be represented as an infinite number of propagating plane waves with appropriate complex weighting. This interpretation can easily be seen from the equation for a single propagating plane wave with frequency ω_0 and wavevector $\mathbf{k}_0$,

$$p(\mathbf{r}, t) = A_0 e^{-i(\omega_0 t - \mathbf{k}_0^T \mathbf{r})}, \quad (50.4)$$

where A_0 is the plane-wave amplitude. Note that for generality, one can allow A_0 to be complex, but for this discussion it is assumed that A_0 is real. The wavenumber–frequency spectrum of this single propagating plane wave is simply,

$$P(\mathbf{k}, \omega) = A_0 \delta(\mathbf{k} - \mathbf{k}_0) \delta(\omega - \omega_0). \quad (50.5)$$

Equation (50.5) shows the Fourier mapping of infinite continuous functions in both time and space to be a point in the wavevector–frequency space.

Now consider a general weighting function of the space–time pressure distribution with a four-dimensional linear shift-invariant filter having an impulse response $h(\mathbf{r}, t)$. Then, the output signal is the convolution of the space–time acoustic pressure signal and the spatial weighting function such that,

$$y(\mathbf{r}, t) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} h(\mathbf{r} - \hat{\mathbf{r}}, t - \tau) p(\hat{\mathbf{r}}, \tau) d\hat{\mathbf{r}} d\tau. \quad (50.6)$$

Equation (50.6) is a filtering operation that can equivalently be represented in the wavevector–frequency domain as,

$$Y(\mathbf{k}, \omega) = H(\mathbf{k}, \omega) P(\mathbf{k}, \omega). \quad (50.7)$$

Equation (50.6) and (50.7) show the direct analogy between the space–time wavevector–frequency representation of spatial filtering and the well-known results from multidimensional linear systems theory. Thus, by using specific spatial weighting functions, one can filter the acoustic field to investigate the propagating directions, amplitude, and phases of plane waves traveling in space. Filtering the spatiotemporal acoustic scalar pressure field is known in the field of array signal processing as *beamforming*. Thus, if one wants to look at plane waves propagating from the direction of the unit vector $\mathbf{k}_0/\|\mathbf{k}_0\|$, one would design a filter such that,

$$H(\mathbf{k}, \omega) = \delta(\mathbf{k} - \mathbf{k}_0) G(\omega), \quad (50.8)$$

where $G(\omega)$ is the desired frequency response to waves propagating in the direction of the wavevector $\mathbf{k}_0$.

Although it may seem limiting to use plane waves as the underlying basis functions for the representation of a sound-field, it should be noted that other orthogonal representations such as spherical and cylindrical basis functions can themselves be represented as a series of plane waves.

The representation of the wavevector–frequency domain is a natural framework for analysis of the spatial and frequency filtering of beamforming arrays, as will be seen in the following.

In spatial filtering by beamforming, one deals with spatial apertures. An aperture is a region over which energy is received. Where an aperture can either be continuous, as in parabolic dishes, or discretely realized as in microphone arrays with multiple microphones, mixtures of continuous and discrete apertures are possible.

In fact, even discrete microphone arrays are a mixture since no microphone is a perfect point receiver. Although sampled aperture systems must generally perform more signal processing, they offer several advantages over continuous aperture systems. A main advantage in using sampled apertures is that they allow for the possibility of using digital signal processing on the individual array signals. Arrays can be electronically steered, can form multiple simultaneous beams, and can be made adaptive purely by electronic means.

50.1.1 Delay-and-Sum Beamforming

Delay-and-sum beamforming, also known as *classical beamforming*, is one of the simplest and oldest techniques for realizing directional array systems. Although not fundamentally limited in bandwidth, early delay-and-sum arrays were used in *narrowband* operation to focus arrays onto a particular point or direction. Since a time delay for narrowband applications can be accomplished with a unique phase shift for each element, narrowband beamforming is typically implemented with phase shifts and is therefore commonly referred to as *phased-array* beamforming. One typical application for microphone arrays is to pick up speech or acoustic signals that are wideband (the desired signals cover many octaves in bandwidth). Therefore most microphone arrays for speech acquisition are implemented as delay-sum beamformers since they are inherently wideband.

Both continuous and discrete beamformers can be seen as implementations of wavevector–frequency filtering as discussed in the previous section. It should be noted that these two beamformers are not mutually exclusive; one can, for instance, have a discrete set of continuous transducers so that the beamformer is a mix of continuous and discrete sampling. Of special interest here is the spatial and frequency response of a finite array of microphones that sample the acoustic field. A classic delay-sum beamformer uses the sum of weighted and delayed samples from an array of N discretely sampled points of the pressure field. In general, a delay-sum beamformer output y is formed as,

$$y(t) = \sum_{n=1}^N w_n s(\mathbf{r}_n, t - \tau_n), \quad (50.9)$$

where the weights w_n are real and the time delay τ_n is applied to the measured signals $s(\mathbf{r}_n, t)$, at microphone n . If the incident field is a planewave with wavevector $\mathbf{k}_0$, amplitude A_0 and frequency ω_0 , then,

$$s(\mathbf{r}_n, t) = A_0 e^{i(\omega_0 t - \mathbf{k}_0^T \mathbf{r}_n)}, \quad (50.10)$$

and

$$y(t) = A_0 e^{i\omega_0 t} \sum_{n=1}^N w_n e^{-i(\omega_0 \tau_n + \mathbf{k}_0^T \mathbf{r}_n)}. \quad (50.11)$$

The real weights w_n scale the signals measured at the positions $\mathbf{r}_n$. Thus, from (50.9), the origin of the term *delay-and-sum* can be seen as simply a description of how the classical delay-sum beamformer is realized.

For a causal delay-sum beamformer, the exponent in (50.11) must be less than or equal to zero for all n . Typically, one element position is selected as the spatial origin. This reference element position therefore defines the vectors $\mathbf{r}_n$. Since the array can be steered to any direction in space and one can select any position to define $\mathbf{r}_n$, an additional delay may be required in order to maintain causality. (Recall that $\mathbf{k}_0^T \mathbf{r}_n$ can be either positive or negative depending on the direction of the incident wave relative to the array.) If an end element is selected as the spatial origin, then the causal delay is equal to the maximum time that an acoustic wave takes to transit the array. Smaller causal delays are also possible depending on which microphone position in the array is chosen as the spatial origin reference and the maximum desired steering angle.

If we set the causal time delay equal to T_0 , then the output for an incident plane wave with angular frequency ω_0 and wavevector $\mathbf{k}_0$ can be written as,

$$y(t) = A_0 e^{i\omega_0 t} \sum_{n=1}^N w_n e^{-i[\omega_0(T_0 + \tau_n) + \mathbf{k}_0^T \mathbf{r}_n]}. \quad (50.12)$$

The maximum plane-wave response corresponds to the direction where the delays τ_n compensate for the propagation delay $\mathbf{k}_0^T \mathbf{r}_n$ of a plane wave propagating with wavevector $\mathbf{k}_0$. This is done by setting the values of τ_n such that these delays offset the time lead (or lag) of a desired direction plane wave propagating over the array. Thus, setting the delays to steer the array to the direction of $\mathbf{k}_0$, means that the last two terms in the exponential in (50.12) cancel out. The delayed and summed output $y(t)$ is now such that all microphone signals propagating from the desired direction are added in phase. This direction corresponds to a maximum amplitude output for any selection of τ_n . Plane waves propagating from directions other than $\mathbf{k}_0$ will result in the addition of position-dependent phase variations, and as a result, the output amplitude will be smaller due to destructive interference.

To summarize, classical delay-and-sum beamforming uses delays between each array element that compensate for differences in the propagation delay of

array beam steering [51.1–4] and a good example of the latter is automatic camera pointing for videoconferencing [51.5–8].

The problem of locating radiative point sources using sensor arrays has long been of great research interest given its theoretical as well as practical importance in a great variety of applications, e.g., radar [51.9, 10], underwater sonar [51.11], and seismology [51.12]. In these applications, source localization is more commonly referred to as direction of arrival (DOA) estimation. A class of celebrated approaches is based on high-resolution spectral analysis, including Capon's minimum variance (MV) spectral estimation method [51.13], and eigenanalysis-based techniques, such as the popular multiple signal classification (MUSIC) algorithm [51.14]. These methods perform statistical fit for DOA with respect to a spatio-spectral correlation matrix derived from the signals received at the sensors. By assumption, the source signal needs to be statistically stationary and narrowband, and the source is located in the far field of the sensor array. However, these basic premises are barely met by speech sources. Moreover, the multipath effect is not taken into account in the formulation of DOA estimation. Therefore, the high-resolution DOA estimation algorithms were found to be incompetent for speech source localization, particularly in reverberant acoustic environments.

For speech source localization, microphone array beam scanning and time delay estimation (TDE)-based localization methods are the two most widely used approaches. A beamformer is a spatial filter that operates on the microphone outputs in order to enhance the signal coming from one direction while suppressing noise and interference from other directions. Therefore, steering a microphone array beamformer and scanning across a room for the highest-energy output leads to an estimate of the direction of an active speech source. Using two arrays and by intersecting the two corresponding direction estimates, the source location

is determined [51.15, 16]. While the beam scanning method has the remarkable advantage that it can easily be extended to the case of simultaneously localizing multiple speech sources [51.17], it has some intrinsic drawbacks. Since speech is a typical broadband signal, ideally a broadband beamformer whose beam pattern would be the same across the entire speech spectrum needs to be developed, which is a technically challenging problem. In addition, most beamforming algorithms assume that the speech sound source is in the far field in order to make the design problem analytically tractable. When the speech source is close to the microphone array, performance degradation can be expected. On the contrary, if the speech source is in the far field, the localization resolution may not be satisfactory as the source direction is only selected from a set of discrete beam scanning angles. The number of beam scanning angles is therefore a trade-off between resolution and computational complexity.

Alternatively, a TDE-based source localization algorithm involves a two-step procedure [51.18]. In the first step, a set of relative time differences of arrival (TDOAs) for different microphone pairs are calculated. In the second step, the acoustic source location is estimated from the TDOAs and using the a priori knowledge about the locations of the microphones. TDE-based source localization algorithms have many merits: they cope well with both narrowband and broadband source signals; their localization resolution can be flexibly adjusted by varying the sampling rate and microphone array size; the effect of room reverberation is only of concern in the first-step TDE and several recently developed robust TDE algorithms are promising for future practical use; and in general they are computationally efficient. Therefore, after continuous investigation over the last two decades, TDE-based speech source localization schemes have become the technique of choice, especially in recent digital systems. In this chapter, an overview of the state of the art of this class of source localization methods is presented.

Depending on the surrounding acoustic environment, there are two signal models for the TDE problem: the ideal free-field model and the real reverberant model. The former assumes no room reverberation while the latter uses an acoustic channel impulse response, usually a finite impulse response (FIR) filter, to describe the effect of room reverberation.

Ideal Free-Field Model

In an anechoic open space, as shown in Fig. 51.1a, the speech source signal $s(k)$ propagates radiatively and the sound level falls off as a function of distance from the source. Then the signal captured by the n -th microphone at time k can be expressed as follows:

$$x_n(k) = \alpha_n s(k - \tau_n) + b_n(k), \quad n = 1, 2, \dots, N, \quad (51.1)$$

where α_n ($0 \leq \alpha_n \leq 1$) is an attenuation factor due to propagation loss, τ_n is the propagation time, and $b_n(k)$ is additive noise. The time difference of arrival (TDOA) between the i -th and j -th microphones is defined as,

$$\tau_{ij} \triangleq \tau_i - \tau_j, \quad i, j = 1, 2, \dots, N. \quad (51.2)$$

The noise signal $b_n(k)$ is presumed to be a zero-mean, white, Gaussian random process, and is uncorrelated with the source signal as well as the noise signals at other microphones.

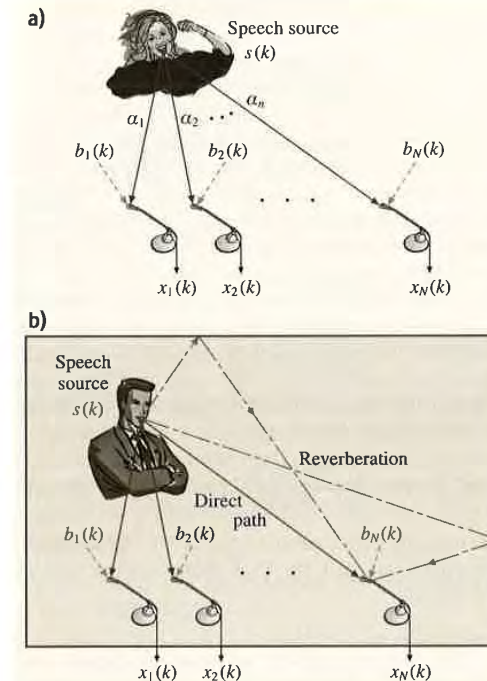


Fig. 51.1a,b Illustration of the two acoustic signal models for time delay estimation: (a) ideal free-field model and (b) real reverberant model

Real Reverberant Model

While the ideal free-field model has the merit of being simple, it does not take into account room reverberation. Therefore, in a real reverberant environment, the ideal free-field model is inadequate and problematic, and we need a more-comprehensive and more-informative alternative to describe the effect of multipath propagation. The real reverberant model treats the acoustic impulse response with an FIR filter, as illustrated by Fig. 51.1b. In such a single-input multiple-output (SIMO) system, the n -th microphone signal is given by

$$x_n(k) = h_n * s(k) + b_n(k), \quad n = 1, 2, \dots, N, \quad (51.3)$$

where h_n is the n -th channel impulse response, and the symbol $*$ denotes the linear convolution operator. In vector/matrix form, (51.3) can be rewritten as

$$\mathbf{x}_n(k) = \mathbf{H}_n \cdot \mathbf{s}(k) + \mathbf{b}_n(k), \quad n = 1, 2, \dots, N, \quad (51.4)$$

where

$$\begin{aligned} \mathbf{x}_n(k) &= [x_n(k) \cdots x_n(k-L+1)]^T, \\ \mathbf{H}_n &= \begin{pmatrix} h_{n,0} & \cdots & h_{n,L-1} & \cdots & 0 \\ \vdots & \ddots & \vdots & \ddots & \vdots \\ 0 & \cdots & h_{n,0} & \cdots & h_{n,L-1} \end{pmatrix}, \\ \mathbf{s}(k) &= [s(k) \quad s(k-1) \quad \cdots \quad s(k-L+1) \\ &\quad \cdots \quad s(k-2L+2)]^T, \\ \mathbf{b}_n(k) &= [b_n(k) \quad \cdots \quad b_n(k-L+1)]^T, \end{aligned}$$

where $[\cdot]^T$ denotes the transpose of a vector or a matrix, and L is the length of the longest channel impulse response in this SIMO system.

In contrast to the ideal free-field model, the time delay τ_n in the real reverberant model is an implicit or hidden parameter. Using such a model, TDE can be performed only after the SIMO system is *blindly* identified (since the source signal is unknown), which looks like a more-difficult problem but is fortunately not insurmountable.

51.2.2 The Family of the Generalized Cross-Correlation Methods

The generalized cross-correlation (GCC) algorithm proposed by Knapp and Carter [51.19] is so far the most widely used approach to TDE. It employs the ideal free-field model (51.1) and considers only two microphones,