Sandlin's Textbook of HEARING AID AMPLIFICATION

Technical and Clinical Considerations

Third Edition

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Preface

Bob and I discussed a third edition to his 1990 and 2000 textbook during a many lunches while overlooking the boats and the harbor in Oceanside, California. Having known, travelled, and taught with Dr. Sandlin for many years, I never ceased to be amazed at his inquiring mind, and his insight into matters both professional and clinical. Most of all, I was overwhelmed by his commitments and his humanity. After all, it's not many who, having passed their 80th birthday, would still feel compelled to contribute to his profession by preparing a third edition textbook. Robert E. Sandlin, PhD, my friend and business partner, died on May 3rd in 2012. He would have been 85 in June that year. He died peacefully with his family by his side, and he displayed the same dignity in passing as he did during his entire life. I have never known anyone quite like Bob, and most who knew him no doubt feel the same way. He was a very special person. And, he was a dedicated audiologist throughout his entire life. His death left me with the task of finalizing his last contribution, and I offer it to his memory.

Editing a text is sort of like directing an orchestra. If the performance is terrible, the conductor rightfully bears the brunt of the criticism. If the performance goes off without a hitch and the arrangement is reasonable, the contributors, be they strings and horns or virtuoso audiologists, should get the credit. Lots of exceptional audiologists, engineers, and clinicians contributed again to this third edition and, if this production is successful, the credit belongs to them.

Some of the chapters in this third edition have not been revised to any extent. Likely, we would all agree that the rules of physics and acoustics, the history of amplification efforts, and so on have not changed all that much in the past 25 years. Also, some of the second edition chapters are no longer in the text. Some of the past contributors are no longer practicing, and some have died. Some chapters have simply been updated, while others have undergone major revision. Even though physics and acoustics have changed little, the technology and clinical approaches to understanding and aiding hearing loss have changed dramatically. There are added chapters that discuss amplification with children, new and future technologies, and practices recommended by professional organizations. We have shortened some chapters that we believe are better managed in topic-specific texts and writings. Overall, we have attempted to gather old and new authors who can address some of the relevant topics (circa 2013) concerning the clinical application of hearing amplification.

It is tempting to think that the entire topic of amplification for the hearing impaired can be covered in one textbook. In practice, and in discussions over our lunches, we found it necessary to eliminate many topics of considerable interest in order to keep the text at a weight that could be managed by most adults using a backpack. I am not so sure we succeeded because, as we reviewed our decisions, we were dismayed in that many topics seem missing. But, we have given it a good start, and the invitation is open to those who follow to fill in the voids we missed.

This preface to the third edition is intentionally short in the hope that its shortness will inspire some to remember the contributions Sandy made to audiology in general and hearing aid practices specifically. My hope, and I believe it was Bob's hope also, is that all readers will find knowledge in the text that will help them deliver better care to their hearing-impaired patients. All of these writers offer their contributions as a tribute to Bob. My gratitude goes to all of them. Finally, it has been my privilege to have been associated with my long-time friend and colleague, Sandy. I hope that all you who read this text will find a similarly rewarding friend and collaborator.

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Summer, 2013

CHAPTER 6

Use of Directional Microphone Technologies to Improve User Performance in Noise

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OUTLINE

INTRODUCTION

One of the most common complaints of listeners with hearing loss is difficulty understanding speech in background noise (e.g., Takahashi et al., 2007). Early studies comparing speech recognition performance of listeners with normal hearing to that of listeners with hearing loss have indicated that those with hearing loss require a better SNR than those with normal hearing (e.g., Dirks, Morgan, & Dubno, 1982; Dubno, Dirks, & Morgan, 1984; Duquesnoy & Plomp, 1983; Plomp, 1978; Schum, Matthews, & Lee, 1991). Evidence also has suggested that the detrimental impact of noise on speech recognition performance worsens as hearing deteriorates (Nabelek, 1982; Nabelek & Mason, 1981).

Considerable research and development efforts have been devoted to hearing aid technologies to reduce the detrimental impact of noise, both in processing-based and microphone-based applications. Processing-based techniques (i.e., single-microphone digital noise reduction algorithms) analyze the incoming signal and then alter the gain/output characteristics to enhance speech and/or attenuate noise. Microphone-based techniques (i.e., directional (DIR) microphone technologies) utilize the outputs from multiple microphones to achieve spatially dependent sound sensitivity, thereby improving SNR. Compared with processing-based techniques, DIR technologies seem to hold the most potential for enhancing speech intelligibility by improving the SNR in at least some listening situations. Specifically, DIR microphone hearing aids are designed to be more sensitive to sounds arriving from certain directions (usually in front of the listener) and less sensitive to sounds from other directions (e.g., from behind the listener). If hearing aid users can place the talker of interest in the front and position the noises to the back, DIR technologies can improve SNR, thereby enhancing speech intelligibility.

For many years, DIR technologies were considered a special feature and were only available in high-end hearing aid models. Today, manufacturers offer DIR technologies in most of their hearing instruments, including entry-level devices. Although DIR technologies have also been implemented on other hearing devices such as cochlear implants (e.g., Chung, Zeng, & Acker, 2006) and bone-anchored hearing aids (e.g., Oeding, Valente, & Kerckhoff, 2010), this chapter focuses on traditional ear-worn DIR microphone hearing aids.

HOW DIRECTIONAL TECHNOLOGIES WORK

Omnidirectional Microphone System

To understand how a DIR microphone system works, it is first necessary to understand an omnidirectional (OMNI) microphone system. A typical hearing aid OMNI microphone system is a closed box that is divided into two small chambers by a thin diaphragm (Figure 6–1). Sounds enter the small microphone port and travel into the chamber. The changes in sound pressure cause the diaphragm to vibrate. The vibration of the diaphragm creates a small electrical signal, which is then processed by the electronic circuit of the hearing aid. OMNI microphone systems are equally sensitive to sound pressure arriving from all directions.

Directional Microphone System: First-Order

The sensitivity of the DIR microphone system depends on the origin angle of sounds. This spatial difference in sensitivity is termed "directivity." There are several ways to achieve directivity. Two designs are commonly used in hearing aids: the single-cartridge design and the dual-microphone design.

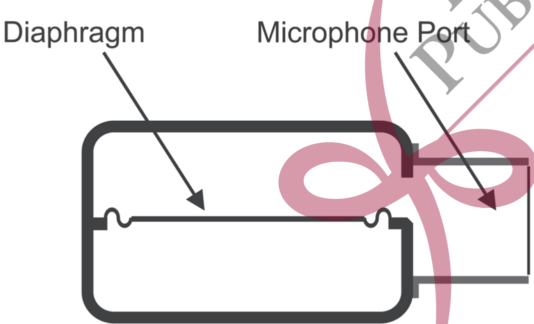

Figure 6–1. The schematic of an OMNI microphone system. Courtesy of Knowles Electronics.

Figure 6–2 shows the schematic illustration of the single-cartridge design. The microphone has one diaphragm housed in a cartridge with two ports allowing sounds to enter on both sides of the diaphragm. The two ports are typically arranged anteriorly-posteriorly on the horizontal plane of the hearing aid as it is placed on the listener's ear. Along the line passing the two microphone ports, the direction in front of the listener is typically defined as 0° and is referred to as the *on-axis* direction. All other directions are considered *off-axis*. Note that there is a mechanical screen, which serves as an acoustic resistor, covering the pathway of the real microphone port.

Now assume a sound source is behind the hearing aid (i.e., 180°). The sound enters the rear port first and then, with a traveling time delay, the front port. This time delay is called *external delay*. The sound entering the rear port is further delayed by the mechanical screen before it enters the chamber. This delay is termed *internal delay*. When the internal delay is set to be equal to the external delay (by choosing an appropriate acoustic resistor), the sound entering the rear port and the sound entering the front port will hit the two sides of the diaphragm simultaneously, cancelling out any movement of the diaphragm. As a result, the microphone will generate no

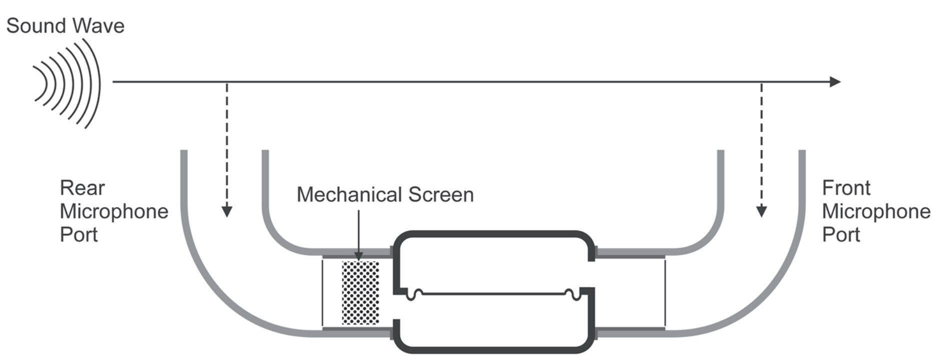

Figure 6–2. The schematic of the single-cartridge design of a DIR microphone system.

output because the sound entering one port is subtracted from the sound entering another port. For sounds arriving at the hearing aid from the on-axis direction (0°), the external delay time will be different from the internal delay. As a result, the sounds entering the two ports will not hit the diaphragm simultaneously, resulting in diaphragm vibration and microphone output.

Compared with the single-cartridge design, the dual-microphone design is more commonly used in today's hearing aids. In the simplest dual-microphone system, the spatial sensitivity is achieved by combining two OMNI microphones (Figure 6–3). The acoustic signal is first converted to an electrical or digital signal by each microphone, and then the output signal of one microphone is subtracted from that of the other. Before subtraction takes place, a delay (i.e., internal delay) is applied to the signal from the rear microphone, but now the delay is achieved electronically or digitally. If the internal delay is set equal to the external delay and the sound is arriving from 180°, output signals from the two microphones will then be identical before subtraction. After signal subtraction, there will be no output from the DIR microphone system.

Although the single-cartridge design differs from the dual-microphone design in structure, these two designs are conceptually equivalent: both rely on the appropriate time difference of the sounds entering the microphone ports and signal subtraction to achieve directivity. This approach is often referred to as *delay-and-subtraction processing*. Because the output of the system is obtained after *one step* of signal subtraction between the sound pressures at the two sides of the diaphragm or between the electronic/digital signals of the two microphones, these systems are classified as first-order DIR systems.

The biggest advantage of the dual-microphone design over the single-cartridge design is the flexibility. Specifically, because the signal delay and subtraction can be achieved digitally, the directivity of the dual-

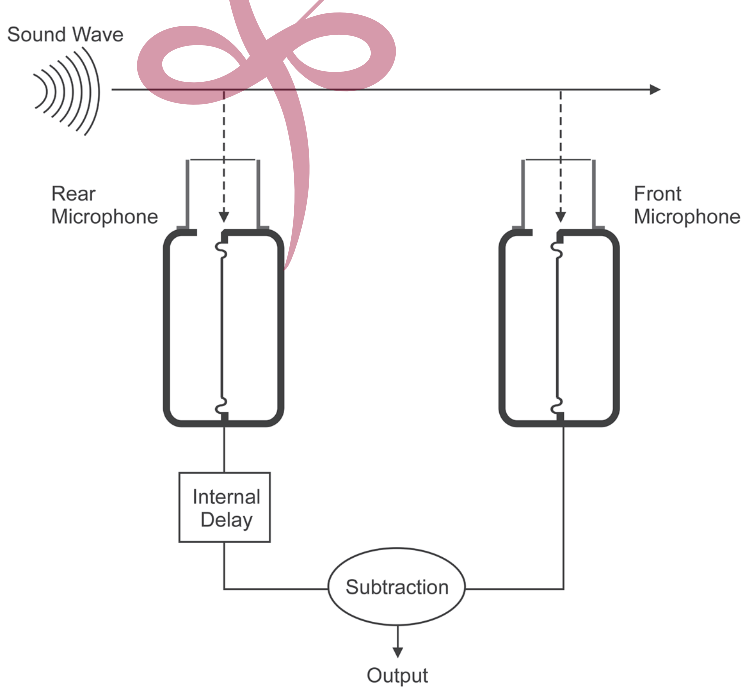

Figure 6–3. The schematic of the dual-microphone design of a DIR microphone system.

microphone design can be easily manipulated. For example, setting the internal delay to be equal to the external delay will, as mentioned previously, attenuate the sound arriving from 180° the most. If the internal delay is programmed to be zero, the microphone system will be less sensitive to sounds arriving from the side of the hearing aid (90° or 270°). This flexibility is the basis of the adaptive DIR system (see the section *Adaptive and Automatic Features*). On the other hand, even though the single-cartridge design can only achieve one directivity pattern (the internal delay is fixed by the acoustic resistor chosen by the manufacturer), this design is free from the potential problem of directivity degradation caused by microphone aging. Specifically, the two OMNI microphones in the dual-microphone design need to be precisely matched in their intensity and phase characteristics in order to achieve and maintain directivity. The dual-microphone system could lose its microphone match over time, thus degrading the directivity. This issue is addressed in the section directivity degradation.

Directional Microphone System: Higher-Order

Directivity can be achieved by combining three microphones. For example, a second-order DIR system has three matched OMNI microphones and uses two steps of signal subtraction between the microphones to achieve directivity (Figure 6–4). The second-order system has been implemented on a commercial hearing aid (Powers & Hamacher, 2002) and is shown to have better SNR improvement than the first-order system (Bentler, Palmer, & Dittberner, 2004; Bentler, Palmer, & Mueller, 2006).

Microphone arrays that consist of multiple microphones have been under development for several years. The microphones in an array can be arranged in any configuration. The simplest configuration is the *broadside array*, in which microphones are placed along the axis perpendicular to the direction of the sound source of interest (e.g., along the listener's forehead using an eyeglass or headband), and the *end-fire array*, in which

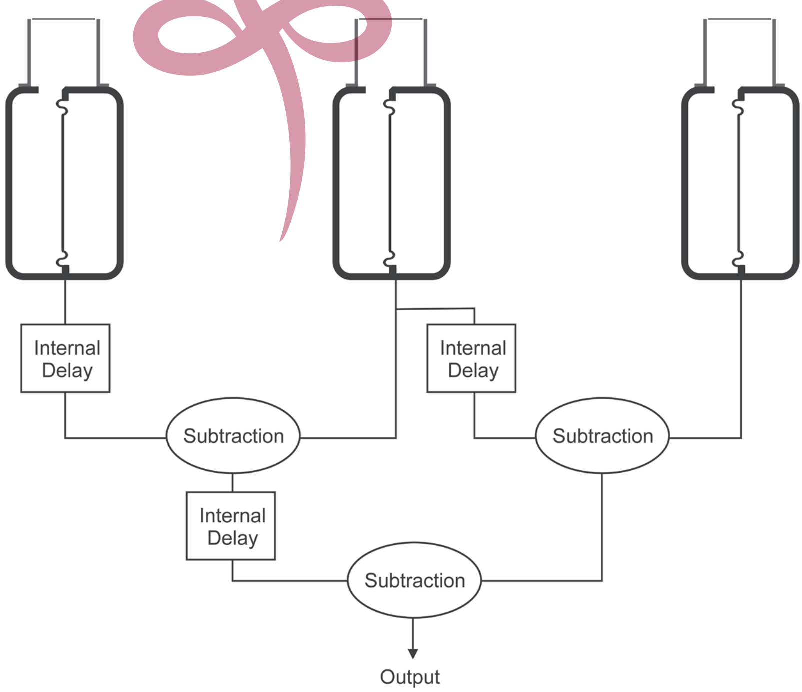

Figure 6–4. The schematic of a second-order DIR microphone system.