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# TECHNOLOGIES FOR SIGNAL PROCESSING HEARING AIDS

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## ABSTRACT

Hearing aids that are currently in use can be subdivided into three basic types. In a traditional linear hearing aid, the gain of the instrument is adjusted manually and remains fixed regardless of signal level. In a compression hearing aid, gain is dependent on signal levels. In an automatic frequency response (AFR) hearing aid the frequency-gain characteristic (i.e., both gain and frequency response) are dependent on signal level. Hybrid circuitry in which a digital unit controls analog audio components is being used increasingly in modern hearing aids. This technology allows for the hearing aid to be programmed by an external cumputer. There are many different kinds of experimental hearing aids. These include complex multi-channel compression amplifiers as well as systems with advanced signal-processing capabilities for noise/reverberation reduction, feedback control and novel techniques for enhancing those features of the speech signal that are either inaudible or misheard as a result of the hearing impairment.

## I. MODERN HEARING AIDS

Until recently, almost all hearing aids were of the traditional linear type. Major considerations in the design of these hearing aids were the frequency-gain characteristic, maximum power output, internal noise and distortion, power consumption and the physical size of the instrument.

Researchers concerned with the fitting of linear hearing aids focussed on the problem of selecting the most appropriate frequency-gain characteristic and maximum power output of these instruments. As a result of these efforts, many different methods have been developed over the years for selecting the frequency gain characteristic and maximum power output of a linear hearing aid [1,2,3,4]. An inherent problem in selecting these characteristics is that the best frequency-gain characteristic for one listening situation (e.g., conversational speech in a noisy room) is not necessarily appropriate for other listening conditions (e.g., in quiet). Compromises need to be made in the selection of frequency-gain characteristics and, to a lesser extent, maximum power output. As a consequence, a traditional linear hearing aid does not provide the most appropriate amplification at all times. The recent introduction of hearing aids with programmable memory represents a new approach to this problem. These instruments allow the user, at the press of a button, to change the electroacoustic characteristics of the hearing aid as needed.

Engineering efforts to reduce the size and power consumption of hearing aids have resulted in instruments that are small enough to fit entirely within the ear canal. Power consumption is also extremely low so that batteries of very small size can be used for several days before needing to be replaced. A point has now been reached where these is little advantage, from a cosmetic perspective, to further reductions in physical size. Engineering development has now begun to focus on improving the signal processing capabilities of hearing aids. This trend coincides with the development of micro-miniature digital circuits, and the use of digital techniques in modern hearing aids is growing rapidly. Many modern hearing aids now use hybrid technology in which a digital unit controls analog audio components. This technology allows for the hearing aid to be programmed by an external unit. Several companies have developed special-purpose programming units for their hearing aids while others have relied on the use of a general-purpose small computer with a custom interface for programming their products. This technology also allows for highly efficient computerized techniques, such as the adaptive paired-comparison procedure, to be used in the prescriptive fitting of programmable hearing aids [5,6].

Compression amplification is the most common form of signal processing in so-called advanced signal processing (ASP) hearing aids. There are three basic forms of compression amplification, automatic volume control (AVC), syllabic compression and compression limiting. In an automatic volume control hearing aid the gain of the amplifier is changed slowly so that long-term variations (> 200 msec) in input signal level that are above the compression threshold produce smaller variations in the output signal level. In a syllabic compression hearing aid, the time constants are relatively short so that the gain of the system varies with the individual sounds of speech. In a compression limiting hearing aid, signals are not compressed until a relatively high signal level is reached. Once the compression threshold is reached, however, a substantial amount of compression is used so that any further increase in input level produces only a marginal increase in output level.

Of the three forms of compression, automatic volume control has been found to be effective in reducing the dynamic range of speech so as to match the limited dynamic range of the impaired auditory system without major reductions in overall sound quality [7]. Compression limiting has proven to be effective in controlling the maximum output of a hearing aid and is superior to peak clipping, the traditional form of output limiting, in that considerably less distortion is generated when output limiting takes place. Syllabic compression is attractive from a theoretical perspective in that the intelligibility of the weaker sounds of speech should be increased, but practical implementation of this form of compression have yet to show improvements in intelligibility that are significantly greater than those obtained with compression limiting [8].

The earliest compression amplifiers were single channel devices in which the total signal power was used to determine the gain of the amplifier. Villchur [9] has pointed out that single-channel compression can actually reduce the level of important high-frequency cues of low amplitude when the low frequency components of the speech signal are relatively powerful. He recommended the use of multi-channel compression in which different compression amplifiers are used for different frequency regions.

Multi-channel compression has been evaluated extensively over the past 20 years with mixed results. Some researchers have reported significant improvements in intelligibility [10,11,12] while others have not [13,14,15,16]. Studies showing improved intelligibility are subject to the criticism that the control condition was not the best possible single-channel system and studies showing no significant improvement in intelligibility have been criticized for using experimental conditions that are not sensitive to the effects of compression amplification. The recent results obtained by Kollmeier, et al [17] are probably the most representative in that significant improvements in sound quality were obtained with some subjects showing a small improvement in intelligibility and others showing a decrement in intelligibility.

A form of signal processing which can be viewed as a special case of multi-channel compression is that in which the frequency response of the amplifier varies with signal level. Automatic frequency response (AFR) hearing aids can be characterized in terms of which spectral regions receive additional amplification as a function of signal level. For example, in a TILL amplifier the treble is increased with signal level, while in a BILL amplifier the bass is boosted with level [18]. Amplifiers of this type are being used increasingly in modern hearing aids. There are large individual differences, however, in the relative effectiveness of the different types of AFR hearing aids (or other advanced signal-processing hearing aids) and it is important that these instruments be prescribed appropriately.

## II. SIGNAL PROCESSING FOR NOISE REDUCTION

People with hearing loss have particular difficulty understanding speech in noise and the focus of much research has been on reducing background noise in hearing aids. Methods of noise reduction can be subdivided into two broad groups, single-microphone techniques and multi-microphone techniques. Hearing aids have traditionally used a single-microphone input and until recently, research on noise reduction for hearing aids concentrated on the single-microphone case.

An important consideration in the development of noise reduction techniques for hearing aids is that the amplification process raises both the speech and noise signals to relatively high sound pressure level. Since the masking effects of noise are greater at high sound pressure levels, one approach to noise reduction in hearing aids is to reduce output levels in those frequency regions where the noise level is high (exceeding that of the speech signal) so as to reduce spread-of-masking effects. An adaptive frequency-dependent method of gain control is needed in order to take temporal fluctuations in the short-term speech and noise spectra into account.

Experimental evaluations of AFR hearing aids designed for this application have not yielded the anticipated improvements in intelligibility for background noises typically encountered in everyday hearing-aid use [19]. Small improvements have been observed under laboratory conditions, however, using specially selected noises, such as an intense low frequency narrow band of noise [20].

A more effective method of single-channel noise reduction is that of spectrum subtraction [21,22] in which the noise spectrum is estimated during pauses in the speech and then subtracted from the speech-plus-noise spectrum. Although some hearing-aid users have expressed a preference for listening to noisy signals processed in this way, objective measures of intelligibility have not shown any improvement in intelligibility [23]. Spectrum subtraction can be quite effective in improving speech-to-noise ratio for voiced speech features and this technique has yielded significant improvements in intelligibility when used as a front end for a feature extraction cochlear prosthesis [22,24].

A method of single-microphone signal processing which offers some hope for small improvements in speech intelligibility is that in which spectral peaks in the speech signal are enhanced. Improvements in intelligibility equivalent to a 4.2 dB increase in speech-to-noise ratio have been obtained when this technique is combined with amplitude compression [25].

Multi-microphone methods of noise reduction have been found to be far superior to single-microphone techniques. Extremely good results have been obtained using adaptive noise cancellation in which one microphone is placed at the noise source [26]. Unfortunately, for hearing-aid applications it is not usually convenient to place a microphone at the noise source and variations of the adaptive noise cancellation technique have been tried with microphones mounted on the head. Small to moderate improvements in intelligibility have been obtained using this approach, depending on the acoustical characteristics of the room and the locations of the speech and noise sources [23]. This technique, however, does not work well with more than two noise sources or if the room is highly reverberant [26].

Adaptive noise cancellation can be used effectively in reducing acoustic feedback (the acoustic feedback signal being treated as noise). Using this form of feedback control, it is possible to increase the gain of a behind-the-ear hearing aid by 8 to 10 dB before the onset of uncontrolled feedback oscillations [27].

The adaptive noise-cancellation technique with two head-mounted microphones has properties similar to that of a directional hearing aid. Even greater improvements in directionality can be obtained using adaptive beam forming techniques [28,29,30]. A relatively simple method of improving the directionality of a hearing aid is to place one or more arrays of microphones on the frame of a pair of eyeglasses. Soede, et al [31] have reported improvements of approximately 7 dB for speech in a diffuse noise field (in comparison with an omnidirectional microphone) for a weighted sum (with appropriate delays) of the microphone outputs.

Perhaps the simplest approach to improving speech-to-noise ratio, with corresponding improvements in both intelligibility and sound quality is to place the microphone at the speech source. This is not always possible, but it can be practical in certain situations, such as in a school classroom. FM and infrared sound transmission systems have been used for this purpose for some time, but the bulkiness of the FM (or infrared) receiver and the inconvenience of switching the microphone to other talkers have limited the applicability of this approach. The recent development of behind-the-ear hearing aids with a built-in FM receiver represent an important step towards improving both the flexibility and convenience of FM signal transmission.

Hearing aid users also have particular difficulty in understanding speech in a reverberant room. The development of signal processing algorithms for reducing room reverberation is an even more difficult problem than that for noise reduction. The results obtained thus far are similar to those for noise reduction. Improvements in sound quality, but not speech intelligibility, have been obtained using single-microphone techniques [23]. Moderate improvements in both intelligibility and sound quality have been obtained using directional microphone arrays in moderately reverberant environments [17]. The latter approach, however, is of little value in a highly reverberant room.

## III. A SPEECH RECOGNITION HEARING AID

A completely different approach to signal processing for hearing impairment is to use automatic speech recognition (ASR) to first recognize what was said, and then to synthesize the speech so as to make it more intelligible. This approach is limited by the current state of the art in automatic speech recognition, but it offers interesting and exciting possibilities for the long term.

In a pilot study of an experimental speech recognition hearing aid, the intelligibility of the system was less than that obtained using a conventional hearing aid, but the experiment did identify the potential advantages of the technique [32]. Specifically, speech in noise can be re-synthesized without any background noise, provided the ASR device can operate reasonably reliably in the noisy environment. Also, the use of speech synthesis rules which exaggerate those features of speech that are often misheard can improve intelligibility, particularly for severely hearing-impaired listeners. This approach may be particularly useful for modifying the acoustic characteristics of speech in ways that cannot be achieved by conventional methods of amplification such as, for example, adjusting vowel duration so as to improve recognizability of the following consonant [33].

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