MULTICHANNEL COMPRESSION IN THE NORMAL EAR AND AS A SIGNAL PROCESSING ALGORITHM FOR THE HEARING IMPAIRED

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ABSTRACT

The nature of sensorineural hearing loss (SNHL) indicates that full-range multichannel compression (FRMCC) signal processing will help the hearing-impaired. Our recent studies of speech perception in hearing impaired subjects support the value of FRMCC with at least 8 channels, especially when the signal-to-noise ratio (S/N) is low. Results from other laboratories, however, have been less favorable. The particular acoustic conditions used in these experiments, plus the restricted time subjects have to acclimatize to each signal processing algorithm, seem to account for the differences among studies, but field testing of FRMCC is needed. Developments in digital signal processing (DSP) have made it possible to plan extensive field tests of binaural 8-channel FRMCC. Hearing-aid users will be able to evaluate the FRMCC in all the acoustic environments they normally encounter, both during and after full acclimatization to the signal processing.

1. INTRODUCTION

The purpose of this report is to provide a simple model for the common SNHL that comes with age and/or noise exposure and to discuss the signal processing algorithm that is essentially the inverse of this model of hearing loss, FRMCC. There is a triple purpose in discussing SNHL in this way: (1) to provide a good first approximation description of the hearing loss, (2) to describe FRMCC and the logic behind its development, and (3) to discuss the problems of testing a signal processing algorithm for application to hearing aids. A critical question in this context is why experimental tests of FRMCC have not more clearly demonstrated its value for improving speech perception in hearing-impaired subjects. Although we should not expect FRMCC to convert the hearing-impaired person into a normal-hearing person, we would expect FRMCC to be better than other amplification schemes whose design seems to ignore the nature of the hearing impairment. This mismatch between expectation and observation suggests that there are problems either in the understanding of SNHL, in the experimental methods, and/or in our interpretation of the results. Resolving problems in any of these areas is important for applying FRMCC, as well as other signal processing algorithms, for the benefit of hearing-impaired patients. A brief discussion of the nature of speech-perception problems in SNHL is also included to facilitate the application of signal-processing solutions to these problems of speech perception.

2. SNHL and FRMCC

2.1 SNHL Model

Figure 1 illustrates a simple model for SNHL that can be useful in understanding the complexity of designing a hearing aid to compensate for the loss [1][2][3]. This particular example is a very common type of loss among hearing-aid users, a sloping high-frequency hearing loss. The impairment is modeled as attenuation located prior to the transduction of sound energy to neural response, with the attenuation at threshold response given by the top, solid curve in the figure. The curve shows the attenuation required at each frequency to give a normal-hearing subject this threshold impairment, ranging from less than 10 dB at 125 Hz to over 60 dB at 4-8 kHz. The values plotted in this threshold curve are the same as those of the standard audiogram, but the shape of the curve would be inverted because higher threshold values are usually lower on the ordinate in the audiogram. None of the other curves correspond to measurements made in the typical audiological examination and the methods by which they are measured will not be discussed in detail here.

As the level of the response increases (successively lower curves in the Fig. 1), the required attenuation to yield a normal response not only decreases, but decreases differentially at different frequencies. Thus, although the sensitivity of the system to different frequencies varies by over 50 dB at the lowest response level (top curve), the sensitivity is near normal and varies across frequency by less than
5 dB at the highest response level (bottom curve). In effect, there is a hearing loss only at high frequencies and low intensities; at high intensities and/or at low frequencies, the response magnitudes are essentially normal (0-20 on this scale corresponds to the range of thresholds for normal hearing). In a different type of patient who had a greater threshold elevation at low frequencies, there would be a low-intensity hearing loss at all frequencies but the high-intensity responses would remain essentially normal at all frequencies.

A further complexity not illustrated in the figure is the frequency specificity of the attenuation: At frequency differences around 10%, attenuation is essentially independent. Thus, the normal response to low-frequency components of a complex sound, and the severely attenuated response to high-frequency components of the sound, can occur at the same time.

Of course, this model should not be taken completely literally because the hearing loss is not the installation of a complex attenuator, but rather frequency-specific damage to a bank of narrow-band compressive amplifiers—the Outer Hair Cells (OHCs) of the cochlea (see [4] for review). Although we did not know about OHC function when Villchur first worked on this model of hearing impairment [1], we now know that the model would be more literally correct if the label on the ordinate of Fig. 1 were changed from "Attenuation" to "Loss of Amplification". But changing the label on the ordinate, would not change the problem attempts to find evidence that mild compression interferes with hearing-aid design. To compensate for this hearing loss, the amplification of each component of sound must depend on frequency and on the intensity at that frequency, but not on the intensities at other frequencies.

One final consideration in this model for SNHL is the time constant of the attenuation variation or, perhaps more correctly, of the compression of the normal OHCs. The time constant is of interest here, primarily to help determine the parameters of the FRMCC or other signal processing that might be able to compensate for the reduction or loss of OHC function. In this context, the OHC-compression time constant is very short, allowing virtually instantaneous adjustment, but apparently without the distortions a compressive amplifier with such short time constants would produce in the auditory frequency range. The position of the OHC within narrow-band subsections of the sound-energy-to-neural-response transduction process may permit these seemingly contradictory properties. In an external signal processing algorithm, however, time constants less than a few ms should produce too much distortion. From the speech-signal perspective, time constants shorter than a few ms also would not be required: Time constants of a few ms will be short enough to permit amplification adjustment of the various frequency bands of single phonemes (vowels or consonants) as they occur in the flow of speech—presumably giving the hearing-impaired listener as good a chance as possible to identify each phoneme.

2.2 FRMCC

The model of hearing loss summarized in Fig. 1 has motivated research on FRMCC, primarily because FRMCC is the signal-processing inverse of this SNHL model. In FRMCC, the input signal is divided into a number of frequency bands and then the signal within each band is amplified with maximum amplification at the lowest intensity and gradually decreasing amplification as the intensity increases. In a true full-range algorithm, the amplification at normal-hearing threshold in each frequency band is enough to bring the intensity of that frequency band up to the impaired threshold and then the amplification is reduced by a constant factor (the compression ratio) until there is no amplification at some high level, e.g., the maximum comfortable intensity in that band. The effect in each frequency band is to map the range of intensity available in normal hearing into the range of intensity remaining in the hearing impairment.

Applying compression throughout the intensity range, as in full-range compression, has two advantages: (1) It corresponds to our understanding of the deficit, as illustrated in Fig. 1. And (2), it permits the use of smaller compression ratios, so that the compression itself will not interfere with speech perception [5]. In contrast to full-range compression, compression limiting uses high levels of amplification with no compression at low intensities and large compression ratios at high intensities, turning our understanding of the hearing impairment upside down and limiting the range of potentially useful intensity information that can be transmitted to the hearing-impaired listener. The rationale for compression limiting has been that any compression would be devastating to speech perception, and thus that its use should be restricted as much as possible. This rationale is weakened by failed attempts to find evidence that mild compression interferes with speech perception (see [5] for further discussion).

In his pioneering studies of 2-channel FRMCC, Villchur [6] did not attempt to measure attenuation functions, as in Fig. 1, for individual hearing-impaired subjects and later experiments confirm the lack of need to do so. Barford [7] measured equal-loudness contours across frequencies for his subjects and concluded that growth-of-loudness functions measured with an entirely different method (calculated from dichotic pure-tone pitch interactions [8]) were equally consistent with straight-line functions. Intensity-response functions measured with an entirely different method (calculated from straight-line functions) in sum, FRMCC should be beneficial for hearing-impaired patients, even without extensive, precise measurements of suprathreshold auditory sensitivity.

3. EXPERIMENTAL DIFFICULTIES

Although the logical argument in favor of the use of FRMCC in hearing aids is very strong, it has been quite difficult to obtain supporting experimental evidence for the efficacy FRMCC. This experimental difficulty is of general interest in the context of signal processing in hearing aids because most of the specific problems are not unique to testing FRMCC signal processing. One set of problems stems from the scientific preference and/or the technological necessity to test complex signal-processing algorithms under a limited range of well-controlled laboratory conditions. Such limited-range testing may be particularly ineffective for an algorithm like FRMCC because its principal logical advantage is that it will maintain near-optimal performance in the broadest range of rapidly changing acoustic conditions. Furthermore, any chosen
set of test conditions may have an unintended bias for or against a signal-processing algorithm. Indeed, such an unintended bias may have had a major effect on research and development of FRMCC with more than a few channels.

3.1 FRMCC, Number of Channels, and S/N

The first major study of FRMCC that included a large number of channels [10] compared the speech recognition of hearing-impaired subjects with 16-channel compression and frequency-shaped linear amplification. The speech was presented at high signal-to-noise ratios (S/Ns) and at high intensities (so that frequency components of the speech remained above threshold as much as possible for the linear amplification). Under those conditions, speech recognition with the 16-channel algorithm was a little worse than that with linear amplification. In contrast, major studies of 2-channel compression [11][12][13] have used different methods, where key measurements are made at low S/N, and the results show 2-channel compression to be somewhat better than linear amplification. Together, these results had been interpreted as evidence against the efficacy of FRMCC with so many channels [4][14][15].

Our more recent results [16] provide evidence for an alternate explanation. The same 8-channel FRMCC that is only almost as good as frequency-shaped linear amplification at high S/N is far superior to that linear amplification at low S/N, suggesting that the S/N difference between the 16- and 2-channel experiments may account for all of the performance difference seen across the experiments. Indeed, other results suggest that the 16-channel algorithm would be superior to 2-channel algorithms at low S/N. Direct comparisons of FRMCC with 4, 6, 8, 12, and 16 channels demonstrate improved performance up to 8 channels and the same performance for FRMCC with 8 and 16 channels [9]. Of course, problems can arise in comparing algorithms across studies based on a single parameter of the processing. For example, there are a number of studies of multichannel compression (MCC) algorithms that are not FRMCC algorithms. The critical difference between 6-channel compression-limiting MCC and 8-channel FRMCC is probably not in the number of channels.

3.2 Frequency-Shaped Linear Amplification

Another problematic aspect of limited-range testing concerns the advantage it gives to frequency-shaped linear amplification—a common control-amplification against which a new algorithm is evaluated. Accurate prior knowledge of the stimuli to be presented permits the experimenter to optimize the level of amplification at each frequency for these particular stimuli. Indeed, if the frequency spectrum and intensity of stimuli are very limited (e.g., one voice at one intensity, as in [7]), then the simple linear amplifier gains an almost-FRMCC-like ability to adjust to the different stimuli across experiments. Of course, the FRMCC algorithm itself gains nothing from limited stimulus variation, because it adjusts its amplification continuously, independent of long-term stimulus similarity.

The experimenter's adjustment of the linear control amplification offers an explanation for its excellent performance at high S/N, where nothing but an appropriate frequency response prevents the linear amplifier from transmitting most of the speech information, at intensities that are above the listener's elevated thresholds. The fact that this frequency response and level of amplification would be inappropriate for other stimuli, is not a problem here. In the case of FRMCC at high S/N, however, there are three possible sources of less than optimal speech recognition: (1) minor distortion introduced by the signal processing, (2) greater amplification of the lower-level noise, and (3) speech information enhanced by the processing, that the impaired listener is not accustomed to hearing. The situation changes at low S/N where, (1) minimal distortion is masked by the noise, (2) noise will not be enhanced, and (3) any bit of additional information may be the difference between identifying and failing to identify the speech.

3.3 Speech and Acclimatization

One final source of problems in testing hearing-aid signal-processing algorithms should be considered. The vast majority of hearing-impaired patients have no complaint except that they cannot understand speech as well as they could, especially under noisy conditions. As a result, speech is the only stimulus set that can be used in hearing aid evaluations, but speech is a complex, highly-redundant information code. The redundancy of the speech code can be a problem because different individuals can use different cues to identify the same speech sound and hearing loss causes people to shift their cue utilization [17]. In this sense, the hearing-impaired listener is not just a normal-hearing listener with reduced information, but a listener with reduced information who uses that remaining information in a different way. If you could give this hearing-impaired listener all of the information available to the normal-hearing listener, you could not expect all of that information to be used "normally" until the hearing-impaired listener had time to adjust to the change.

Readjustment of cue utilization may be part of the explanation for the months an individual needs to acclimatize to a new hearing aid [18]. Whether or not shifting cue utilization and acclimatization are related, however, both indicate the need for extended experience with a new algorithm before we can measure the maximum benefit a patient would receive. In addition, acclimatization to complex signal processing algorithms may take longer than to the conventional hearing aids for which it has been studied [4]. Highly significant improvements in speech perception have been found for 16-channel FRMCC after extended listening experience in the laboratory [19], but such complex algorithms have not been studied outside of the laboratory. Our field studies of binaural 8-channel FRMCC [20] currently being planned include measurements of acclimatization, and also of specific consonant confusions to indicate changes in cue utilization.

4. FUTURE RESEARCH

Further work on FRMCC, as well as research on other signal processing algorithms for hearing-aids, will have to include field testing as the major evaluation method. Now that portable, programmable DSP units are available, there is no reason to accept the difficulties imposed by restricting hearing-aid evaluation to the laboratory. Of course, some laboratory testing will be necessary to establish that a new algorithm can achieve its goal under the conditions for which it was designed, but then the primary evaluation should be shifted to the field. Field evaluation not only
avoid the types of problems discussed earlier from the history of FRMCC research, but also tests the idea of applying the algorithm to the real problems of hearing impairment. Some signal processing algorithms may solve problems, but at the same time create others in situations commonly encountered by the hearing impaired. It is clearly possible, for example, that FRMCC might create variations in the signals that are too complex for stable acclimatization to occur and thus that it would yield significant recurring phoneme confusions (although previous results [19] suggest otherwise). Similarly, a noise-reduction algorithm that facilitates speech perception in frequency-shaped white noise may interfere when the noise consists of a small number of other voices. For any particular algorithm, the specific advantages and disadvantages will be different. Only with extended field testing can we be confident that the vast majority of such problems will be revealed.

In addition to initial evaluation of algorithm design goals, laboratory testing might be valuable in understanding or verifying problems or successes found in the field. For example, if subjects in speech intelligibility in quiet and in noise produced by two-channel hearing aids had reported from field experience that FRMCC was especially beneficial at low S/N prior to [9], such a follow-up laboratory study would have been useful because the result was unexpected in the context of previous FRMCC theory. Overall, however, the primary focus of future hearing-aid research must be shifted from laboratory study to field testing.

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REFERENCES


