Present and Future Technology in Hearing Aids

Technologie actuelle et future dans le domaine des prothèses auditives

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Abstract

This paper lists some problems encountered by hearing impaired people and the proposed solutions. It critically reviews those solutions with regard to their theoretical foundations and their applicability in daily life situations. It further questions the introduction of new technology as a major breakthrough when the technology is an application of available general technology for which the benefit for hearing impaired people has not been demonstrated. Some of these breakthroughs are based on miniaturization of the hearing aid for which the audiological consequences cannot be expected to be positive. An approach is offered based on present knowledge of processing in the impaired auditory system. The consequences of the approach for syllabic compression and some results are presented.

Résumé

Le présent document énumère certains problèmes rencontrés par les malentendants ainsi que les solutions proposées. L'auteur examine de façon critique ces solutions en fonction des bases théoriques et de leur application dans la vie quotidienne. Il met en doute l'introduction de technologies nouvelles qui passent pour une découverte importante lorsque ces technologies sont une application de la technologie générale disponible et lorsque les avantages pour les malentendants restent à démontrer. Certaines de ces découvertes sont fondées sur la miniaturisation des prothèses auditives pour lesquelles les conséquences sur le plan audiologique sont peu favorables. Il est question d'une approche basée sur les connaissances actuelles du traitement du système auditif déficient. Les conséquences de l'approche de la compression des syllabes font l'objet de discussions et certains résultats sont présentés.

Introduction

New technology is frequently introduced in commercial hearing aids as a break-through that will eliminate many of the limitations of existing hearing aids. The design is often a technological achievement but not so much an audiological achievement. The technological advance typically is not based on audiological data at all, and proper testing of the audiological claims is not always performed. Thus, the design is based primarily on technological innovation and not on audiological knowledge of hearing impairment.

It seems necessary to get more interaction between technology on the one hand and audiology, phonetics, and related fields on the other. New insights from research in the latter fields should be incorporated in the approach to hearing impairment and in the design of new hearing aids. This need will be even greater in the near future because digital techniques will present us with many new possibilities and will take away some of the usual limitations of controlling only a small number of parameters. More detailed tailoring of the aid to the individual characteristics of the hearing loss of a particular client will become possible.

A better determination is required of the relevant parameters of hearing function as well as of the effects of these parameters on sound and speech perception in noise and in quiet and the effects on the way a hearing aid may improve perception. Unfortunately clinical procedures have not been developed yet. Only after the development of such tests can a good interaction take place between designers of new hearing aids and prescribers.

Complaints of Patients

The major problems encountered by hearing impaired persons are:

1. *Loss of sensitivity*: Low-level sounds are no longer heard. It is measured as the audiometric threshold and is often the only measure considered in rehabilitation (calculation of aided sound field threshold). The threshold may vary with frequency; the loss is generally larger for higher frequencies.

2. *Smaller dynamic range*: The maximum permissable sound level often does not change in cochlear loss. In combination with the hearing loss it results in a much smaller dynamic range for hearing, sometimes even smaller than the range of speech levels found in everyday communication.

3. *Poor speech discrimination, particularly in noisy situations*: Patients with small losses often do not complaint about communication in quiet situations but only about

communication in noisy situations. There is no simple relationship between loss of intelligibility and loss of sensitivity.

4. *Distorted sounds*: It is often reported that sounds are distorted. A variety of reasons have been proposed, for example, diplacusis and effects of processing by wider auditory filters.

Patients often do not complain explicitly about any one of these problems. They note communication problems and get rather nervous and uncertain about situations that are too difficult to handle. The hearing impaired person does not feel comfortable, tries to evade certain situations, gets into problems at work and/or at home, and becomes stressed (Saunders & Haggard, 1989).

Audiological research (for references see Phonetic and Audiologic Criteria) has shown that hearing loss is accompanied by poor spectral and temporal resolution. It might well be that these effects are more relevant to speech perception than the threshold shifts and the loss of dynamic range. In order to understand the significance of these two characteristics, we have to relate them to phonetic features of speech. Before we do so, however, we shall first describe the technical solutions put forward by designers to overcome the encountered problems and the pragmatic solutions used in the clinic.

Technical Solutions

Problems in communication have been tackled in a pragmatic way often without paying much attention to audiological effects. The solutions to the aforementioned problems are discussed below.

Frequency-dependent Amplification

No frequency response characteristic is used that fully compensates for the audiometric hearing loss because of the reduced dynamic range; the amplification is so adjusted that a maximum amount of information relevant for speech is presented above threshold (and below the uncomfortable loudness level). This results in prescription rules like NAL, Skinner and Pascoe, POGO, Lybarger, Berger, half-gain rule, and so forth. The approach is basically linear; the technical implementation may involve active filtering. The rules are based on the threshold shift, sometimes also on the reduced hearing range and on the average spectrum of speech; it sometimes involves the calculation of the articulation index as a measure of speech intelligibility. The use of smaller gains rather than full compensation at threshold indicates the limited or negative effects of compression circuitry on speech intelligibility. Such systems could have solved the problem of recruitment by reducing automatically the gain for high-level input signals.

Compression

The smaller dynamic range often requires resetting of the volume control. Automatic readjustment would be preferable. Compression circuitry has been used in hearing aids for this goal for quite some time. Three goals have been distinguished; the choice between them depends largely on the available hearing range (Walker & Dillon, 1982):

1. *Limiting the output level* in order to prevent hearing damage from sounds that are too loud and to prevent rejection of the aid by often recurring unpleasantly loud sounds. The system should be used if the hearing range is large enough to cover speech of different speakers under different acoustic conditions. The dynamic range should exceed at least 70 dB.

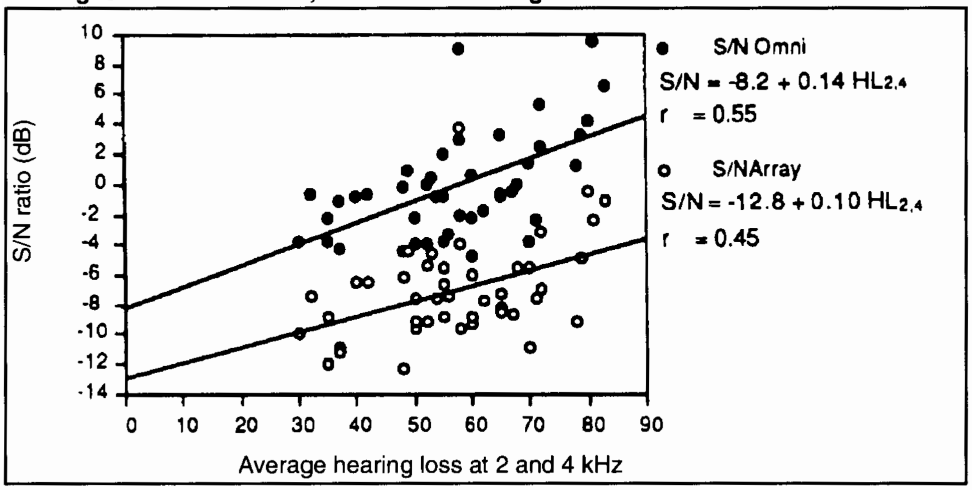
2. *Automatic volume adjustment* to suit different acoustic conditions and different speakers without affecting the dynamics of speech. Speech of one speaker should not be processed by the system. The dynamic range should be at least 50 dB.

3. *Syllabic compression* to reduce the dynamics of speech of one speaker; it should match the dynamic range of speech to that of the hearing loss. It should be used if the dynamic range is smaller than about 50 dB.

The different goals require different settings of the parameters of the compressor, particularly the attack and release times, the compression threshold, and the compression ratio. The requirements often vary with frequency implying that a multi-channel system is necessary. A limiter should be used if the dynamic range is wide and different voices under different conditions do not cause problems. Sudden loud sounds now may cause problems, such as slamming doors. Peak clippers could be used but they distort the sound. The limiter requires a very short attack time, a high compression threshold, and a high compression ratio.

Automatic volume adjustment should be used if the dynamic range is large enough for one speaker but not for different speakers under different acoustic conditions. Speech itself should not be affected and the time constants should therefore be rather long. The compression threshold should be low. Syllabic compression should be used if the dynamic range of one speaker exceeds the hearing range; the speech information itself should be compressed within the hearing range. It requires a fast-acting compressor that can effectively reduce the speech modulations. The threshold should be low

Figure 1. Scatterdiagram showing the S/N ratios for the listening condition with an omni-directional microphone and an microphone array as a function of the average hearing loss at 2 and 4 kHz, and the derived regression lines.



Fourcin stressed the point that a conventional hearing aid might overload the information transfer capacity of an ear with only residual hearing (Rosen, Walliker, Fourcin, & Ball, 1987; Faulkner, Fourcin, & Moore, 1990). He proposed to reduce the amount of amplified information to those features that will ensure maximum speech intelligibility in combination with lip reading. He introduced the SiVo hearing aid and showed advantageous effects for the target group. Field tests in different countries are currently being carried out. The possibilities of speech feature extraction are being investigated. However knowledge is lacking in this field (see for example, Dalsgaard, Fink, Pederson, & Sørensen, 1990). Transposition of information to a frequency range where hearing is better has been tried and found to be largely ineffective, probably because there are no free channels available in the auditory system.

and the time constants short. The effect of syllabic compression on speech perception should be tested in every patient for whom it is used, because we do not know how well the patient is able to process the distorted speech.

To avoid misunderstanding we shall use the words limiter, volume control, and syllabic compressor for the different goals and circuits. Most commercial hearing aids perform some kind of compression. The choice of parameters reflects a compromise between the different possible goals. The rationale expressed explicitly most often is to reduce the dynamic range and to match speech sounds to the dynamic hearing range. However the release time often is rather long and the compression threshold is set at a high level. Tests of the effect of compression on speech intelligibility show generally poor results both in research and in practical applications, although wearing comfort may be improved (Plomp, 1988). Consequently the compressors are not frequently used in the clinic.

Poor Cochlear Processing

There are no systems, yet, that compensate for poor cochlear signal processing. However, many attempts have been made to overcome its effects by improving the signal-to-noise ratio.

Speech Intelligibility in Quiet

Moore (Simpson, Moore, & Glasberg, 1990) proposed a scheme that would compensate for poor frequency resolution: the mexican hat processing. The signal is processed (sharpened in the frequency domain) in such a way that convolution with the poorer frequency resolution of the impaired ear will result in an almost normal frequency resolution. The system should also improve the signal-to-noise ratio. No experimental data are available for the moment.

High-frequency emphasis in hearing aids may improve speech intelligibility in certain patients, particularly those with (steep) high-frequency losses. The goal is to raise the amount of transferred information within the hearing range as calculated by the articulation index (Skinner, Pascoe, Miller, & Popelka, 1982). This processing is in general linear, but a combination of high-frequency boosting and compression may be necessary because of the small high-frequency hearing dynamic range. However, this combination is rarely used in practice because of poor intelligibility results.

Speech Perception in Noise

Speech intelligibility in noise is supposed to improve by the following:

Linear tone control, either automatic or manual. Low frequencies dominate in most background noises (reverberation, fans, engines, footsteps, etc.). The physical signal-to-noise ratio is improved by changing the frequency response of an aid. This fact has been known for many years and some hearing aids were equipped with a switch to reduce low-frequency gain. It is even still in use today. Remote control may promote its use. However, patients tend to find it difficult to judge situations and set the aid accordingly. Automatic tone control is more difficult to realize because it is difficult to provide the aid with criteria to differentiate the wanted low-frequency speech sounds from the unwanted background noises. The emphasis of the high-frequency gain for patients with high frequency noises is important and it has been shown that it can improve the signal-to-noise threshold (Verschuure & van Benthem, 1992).

Directional microphones. A directional microphone preferentially picks up the sound coming from the front (speaker) and suppresses sounds coming from other directions. Directional microphones are available on many hearing aids. Most of them are cardioid microphones. The directionality depends on frequency and is rather limited. Soede (1990) showed it to be about 2.5 dB in KEMAR. It is rather poor in the higher frequency range and depends strongly on the angle between source and noise. Head diffraction also changes it effectiveness. Festen (1984) showed the effect in the concha, including the effects of the pinna, to be about 1 to 2 dB.

Signal processing schemes have been introduced, such as the adaptive monaural beamformer (Peterson, Durlach, Rabinowitz, & Zurek, 1987; Peterson, Wei, Rabinowitz & Zurek, 1990), the adaptive binaural beam former (van Compernolle, 1990), and the adaptive noise canceller (Weiss, 1987; Schwandler & Levitt, 1987). These systems have been shown to be effective in stationary situations and with a limited number of noise sources. The effectiveness in reverberation and in life-like situations with multiple noise sources is poor.

Array techniques also have been proposed (Helle, 1986; Tyler & Kuk, 1990). They seem far more effective and robust. Proper choice of parameters make them effective up to the high frequency range, which in hearing aids is up to about 5 kHz. Soede (1990) found an improvement of the measured signal-to-noise ratio in a diffuse noise field in patients of about 7 dB. Most striking was the effect that the poorer the signal-to-noise ratio of patients was, the larger the improvement was (Fig. 1). The array microphone restored a normal signal-to-noise ratio in most patients. Binaural use of arrays further improved the observed signal-to-noise threshold by about 2.5 dB in patients.

A disadvantage of effective directional hearing aids is the use in traffic and for the detection of warning sounds. The system therefore should also contain a non-directional microphone. There is no information so far about the amount of improvement of the signal-to-noise ratio that is needed by a patient with a certain hearing loss or loss of speech intelligibility in noisy situations. The mexican hat concept discussed earlier could also serve as a signal processor to improve signal-to-noise ratio.

Assistive devices like hand-held microphones, loop-induction systems, FM-systems, and infra-red systems. The goal of these devices is to place the microphone as close as possible to the speakers mouth, thus reducing the relative level of background noise and reverberation. The systems are very effective as can be deduced from their general use in schools for the hearing impaired and in theatres and churches in our country. They require installation and some instruction on their use.

Noise reduction systems (e.g., zeta-blocker). Some of these systems use techniques like directional microphones or adaptive filtering. They don't seem to be very effective in every-day situations (Tyler & Kuk, 1990). The problems of effective noise reduction are:

1. Distinction between unwanted and wanted signals because of great similarity of spectral and temporal characteristics when in a multi-talker environment (Summerfield & Stubbs, 1990): Reverberation even adds to this problem; the speaker becomes his own "noise" source. The advantage of reverberation is its frequency-dependence; filtering out the low frequencies should improve the signal-to-noise ratio.

2. The threshold for speech reception in noise is negative for normal hearing people (Plomp & Mimpen, 1979; Soede, 1990; Verschuure & van Benthem, in press): The systems should be effective at negative signal-to-noise ratios wherein the noise dominates the sound. If temporal and frequency characteristics are not too different for speech and background noise, the distinction between unwanted and wanted sounds is almost impossible. There seem to be no effective schemes yet, except directional microphones, to overcome this problem (compare squelch effect [Markides, 1977]).

3. It is possible to suppress stationary signals by readjusting the gain in a certain frequency channel (van Dijkhuizen, 1991) or by adaptive filtering: Small signal changes, reverberation and head movements may interfere seriously with the effectiveness of such systems.

Internal distortion. There are no systems aimed at reducing distortions because we don't know where and how they are generated: There is some evidence that poor frequency resolution results in distortion. In a meeting of the Dutch Society of Audiology processed speech was demonstrated, which was Fourier transformed, put through a poor filter (bandwidth about 1 octave), and inversely Fourier transformed again. The speech sounded normal if the filter was narrower than about 1 octave, and sounded blurred and distorted and was difficult to recognize if the filter was wider (Keurs, Festen, & Plomp, 1992). It is not certain whether this effect actually takes place in the hearing impaired listener but the observation is suggestive.

Digital Techniques

One of the most challenging new developments is the introduction of digital techniques. At the moment we see:

1. *Remote control of settings* like volume control, frequency characteristic and so on (Quattro, Pharo): The hearing aid is principally an analog system with digital control of some parameters. It is to be expected that such systems will become available for most hearing aids.

2. *Digital control of the tuning* parameters of an analog system: Digital control provides the possibility to tailor the aid to the needs of a particular patient; the Phox-system allows for a better tuning of the frequency characteristic, the PMC system for tuning the frequency characteristic and setting of the AGC circuits. The Quattro makes it possible to adapt the frequency characteristic to different environmental situations among which the patient can choose. The principal advantage of such systems is not a new design but a better tailoring to the patients needs. The systems are useful to prescribers and may reduce the number of hearing aids that are produced and marketed.

3. *The development and design of digital processors*: The size of the aids and power supply limitations have been major obstacles to date. Some of the designs described above have been implemented on digital systems and are not easily implemented on analog systems.

The application in cochlear implant processors is already a reality and the application in commercial aids seems only a matter of time. Real-time digital processing will open up new horizons because the power of digital systems seems greater than we can understand at the moment; the major problem is our lack of audiological knowledge of preferred and desired processing schemes.

We have to realize that high-fidelity procedures do not apply; the aids have to be designed for hearing impaired people, taking their abnormal processing into account. The newly developed Philips DCC system for digital recording on normal compact cassettes shows that a psychophysical approach does work for normal hearing people. The system reduces the bit stream from 16 bit to only about 4 bit by *hiding* distortion products in spectral and temporal ranges where masking (frequency and temporal resolution) makes perception impossible. The system has a very high amount of distortion but the sound quality is extremely good, in fact comparable to CD quality.

Phonetic and Audiologic Criteria

Most designs of nonlinear processing in hearing aids focus on the problem of the smaller dynamic range and noise reduction. Values to describe the dynamic range are usually taken from static determinations of the loudness curve. The measured static recruitment is compensated by automatic gain control. The rationale is that this will result in the restoration of a normal loudness curve (e.g., multifocus system). It is questionable whether static measures are the closest link to the perception of a strongly fluctuating signal such as speech. Other parameters of hearing may be equally or more important.

The effects of altered spectral resolution (upward spread of masking) and poor temporal resolution seem very important. It is necessary however to relate the consequences of these psychophysical effects to properties of speech signals (Verschuure, Dreschler, Haan, et al., 1992; Verschuure, Dreschler, & Haan, 1992).

Simplified Description of Speech

Speech is a complicated fluctuating signal which can be simplified into (e.g., no coarticulation) steady-state and transient parts. Steady-state parts last some hundred milliseconds and differ in spectral contents. They are the (semi)vowels. The energy level is high and their duration is relatively long. Perception is based on the detection of the spectral peaks called formants. Their significance to speech understanding is rather small. It gives the brain time to process and interpret speech. Transient parts last only a few tens of milliseconds, and differ primarily in time structure and crudely in spectrum. They are the consonants and transfer far more information than vowels. The energy level is low. Perception is based on a number of parameters such as voicing, transient character (plosives, fricatives), and dominant frequency region (comparable to formants).

The dynamic range of speech sounds is about 30 dB when speaking in a flat voice and can be about 45 dB with intonation. If we take different voices and voice levels into account, the total range is about 60 dB. The implication is that people with hearing losses larger than about 50 dB HL can get into communication problems when using linear hearing aids.

Audiologic Criteria

We know that the frequency resolution of the hearing impaired ear (as measured in psychophysical and physiological tuning curves) is much poorer than that of the normal ear (Wightman, McGee, & Kramer, 1977; Patterson et al., 1982). This means that low-frequency energy tends to mask high-frequency information, even if the high-frequency information could be detected by the ear in the absence of the low-frequency sound (upward spread of masking). The effects on speech perception are as follows:

1. Reverberation is much longer in the low-frequency range than in the high-frequency range. High-frequency infor-

mation like second formants in vowels and high-frequency consonants are masked.

2. Many background sounds like fans and engines have a lot of low-frequency energy, thus masking the high-frequency parts of speech sounds. The effect explains the problems hearing impaired people experience in noisy situations even with hearing aids.

3. Speech information may even mask itself. A loud first formant may mask a second formant forcing the hearing impaired to use another perceptive field. The effect explains the effects found by Bosman (1989) that hearing impaired people perceive vowels primarily in the first formant-duration plane and not in the first formant-second formant plane. The question remains as to whether this finding shows a different hearing strategy by the hearing impaired listeners or a limitation placed upon them by the choice of frequency response of the aid.

A second problem of the hearing impaired is their poorer temporal resolution (Nelson & Turner, 1980; Zwicker & Schorn, 1982). The reason for poor spectral resolution could be that hearing impaired people tend to listen at a level closer to their threshold. We know (Verschuure, Kroon, & Brocaar, 1983) that at low levels the time constant of the hearing system is about 200 ms, while it is about 20 ms at levels above about 30 dB Sensation Level. This effect means that low-level consonants are easily missed after a high-level vowel. In fact hearing impaired people often mention their problems in perceiving the end of a word.

Implications

Hearing Aid Prescription

The above has implications for hearing aid prescription and hearing aid design. Speech information should not only be above threshold, as in descriptions using articulation index methods and target gain rules, but also be detectable. This implies high-frequency emphasis for the detection of the second formant even though the dynamic range at the high frequencies is often very limited. Strong compression in the high frequencies boosts low-frequency amplification and, depending on the compression parameters, may easily mask essential speech information. Low-frequency losses, as in Meniere's disease, should not be fully compensated. Although the articulation index may get higher, the actual advantage to speech intelligibility may be lost again due to masking effects. Generally, low-frequency gain has to be smaller than high-frequency gain. The splitting-up of a signal into many channels with independent compressors may cause serious problems for the detection of high-frequency information because the entire spectrum may not be perceived, and speech information in some channel may not be detectable.

The importance of the high-frequency response in hearing aids is clear. It should be controlled carefully by using insertion gain measurements. The objections of patients, mostly suffering from high-frequency losses, to the sound quality should be discussed. The patient should be given time to adjust to the sharp and thin sound. Libby horns and aggressive sounding hearing aids should not be ruled out beforehand.

The poorer temporal resolution also has its implications for the detection of overshoots if a compression system is used. Overshoots are easily detectable, while undershoots may be masked. Overshoots should be avoided as much as possible. The usual choice of time constants in hearing aids will lead to large overshoots not reducing transient amplitudes. The reduction in levels is thus smaller than would be expected on grounds of the chosen compression ratio as was found by de Gennaro (1981). If we aim at compensation of the level differences between vowels and end consonants, fast syllabic compressors should be used with release times not exceeding 20 to 30 ms. The time constants of commercial hearing aids are generally far longer, and it can not be expected that they work as syllabic compressors.

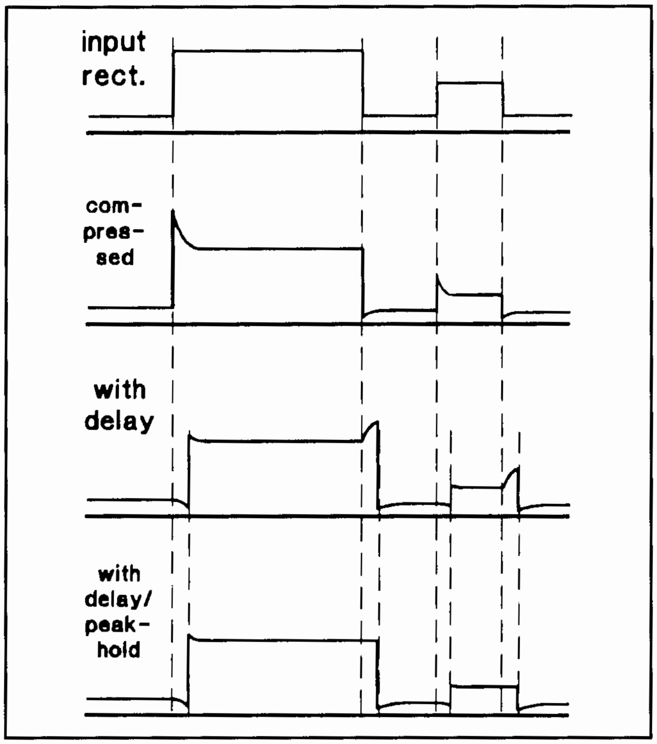
The presence of overshoots and the long time constants in commercial hearing aids makes an advantageous effect on speech perception not very likely. Syllabic compression is just not implemented in hearing aids. The compressors work as limiters and may be effective as such. They also might serve as an automatic volume control that helps to avoid frequent manual readjustment of the hearing aid, although Plomp (1988) argues that longer time constants would be desirable.

The above argument shows the need for methods describing the effective behaviour of compression systems. A modulation technique could serve as such a tool. Verschuure, Maas, Stikvoort, and Dreschler (1992) have used such a technique and they showed that the effectiveness of compressors strongly depended on the chosen parameters.

Hearing Aid Design

New developments to improve the signal-to-noise ratio should be followed carefully. This subject is also important for broadcasting, recording sessions, and computer recognition of speech. Array techniques seem very promising but so do schemes to use binaural processing which should be studied (van Compernolle, 1990). Assessment in hearing aids should include testing on patients.

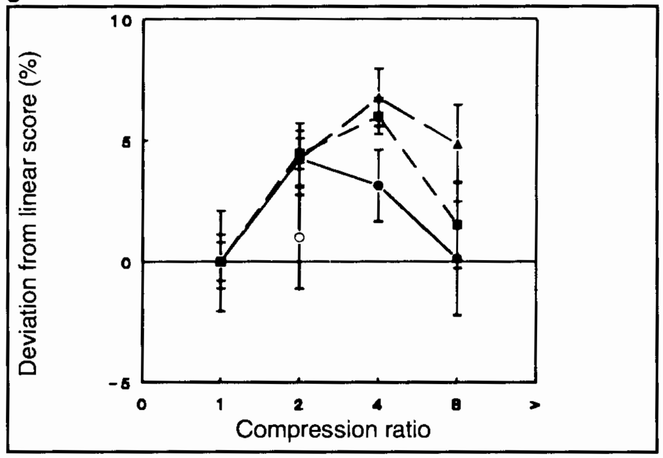
Figure 2. Signal processing for square waves. Overshoots are suppressed by delaying the speech signal in time. The overshoot that would thus appear at the end of the strong signal (vowel) is suppressed by a peak-hold circuit.



All of the arguments under Hearing Aid Prescription have consequences for the design of hearing aids, for tuning the frequency characteristics, and particularly for compression systems. Verschuure et al.(1990), Verschuure, Dreschler, Haan et al., (1992), and Verschuure, Dreschler, and Haan (1992) have studied the effect of compression on the intelligibility of speech in patients with a discrimination loss taking these arguments into account. In a number of studies they determined the effect of a number of parameters. They always measured the intelligibility for compression ratios between 1 (linear) and 8. A delay of the signal was introduced to suppress the overshoot (Fig.2). The resulting overshoot at the end of a stationary signal (vowel) was suppressed by a peak-hold circuit maintaining the reduced gain for some extra ms after the signal had dropped in level. The processing was implemented on a prototype digital signal processor developed by Philips (Stikvoort, 1986) and on a DSP 560001 processor. It was shown that the best intelligibility was found for compression ratios between 2 and 4. For those ratios, the maximum intelligibility was higher than for the linear condition after elimination of presentation level effects.

An example of typical results found with this system is given in Figure 3, in which we present data on 6 patients with a poor speech discrimination (less than 85%) in the linear condition. The figure shows the results for different compression ratios, the compression threshold always being at -50 dB vs maximum input level. The results are given as the deviations from the linear score. The filled circle gives the result for a relatively slow syllabic compression system suppressing modulations up to 14 Hz (attack time of 10 ms, release time of about 30 ms, delay of 7.5 ms, peak-hold time of 11 ms), the open circles for the delay switched off. It shows that without the delay the improvement in speech intelligibility is lost. The results for a faster syllabic compressor system suppressing modulations up to 22 Hz (attack time 5 ms, release time 15 ms, delay of 4 ms, and peak-hold time of 5 ms) are indicated with squares. The triangles represent the results of adding a high-frequency boost in a two-channel system compensating for the full hearing loss just above threshold and limiting the gain at the low-frequency channel for upward spread of masking. We see that the faster system is more favourable, but that the high-frequency boost does not add much to the speech score, perhaps because of over-emphasis of the high frequencies in the linear condition.

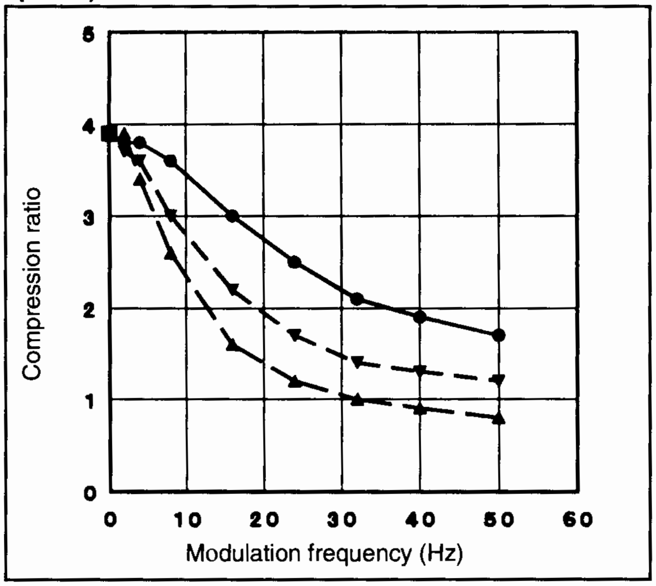
Figure 3. The effect of syllabic compression on speech intelligibility for different settings of the parameters as given in the text.



There are only a few reports in the literature claiming a higher intelligibility with compression than without (Villchur, 1973; Moore, Glasberg, & Stone, 1991) except when averaged over a number of levels. It is particularly striking that Villchur mentioned average time constants that were in the same range as ours. Most other investigators used longer time constants in order to reduce transient distortion.

In the experiments described above (Vershuure et al., 1990; Verschuure, Dreschler, & Haan, 1992), the low-frequency gain was limited by the high-frequency gain in order not to be troubled by upward spread of masking. However, this circuit interferes with the compression. In a later experiment we therefore determined the effect of extra high-frequency gain by introducing a high-pass filter with a slope of 6 dB/oct, either just before the compressor or just after the

Figure 4. The effective compression ratio determined by the amplitude of the first sideband of the spectrum of a modulated sinusoid for different delays of the signal: 3ms (circles), 10ms (triangles downward) and 0ms (triangles upward).



compressor without the anti-upward-spread-of-masking filter. We found better vowel discrimination but poorer consonant discrimination with the filter. The positive effect of compression on speech intelligibility in patients was found only for short time constants. The intelligibility improvement was poorer for attack times of 2 ms and 10 ms as compared to 5 ms. The release time should be shorter than 30 ms. For times shorter than 10 ms the sound gets very distorted. However, we have done no intelligibility experiments for this condition. At the moment we are testing speech intelligibility in noisy situations. The first results indicate no serious damage to the positive effect of compression on speech intelligibility by the life-like background noises.

We can conclude from the results above that indeed much smaller time constants should be used than are realized in commercial hearing aids, provided we can suppress the overshoots and hide the transient distortion in low-level parts of the speech signal. The need for a method describing the effectiveness of a compression system has already been mentioned. Figure 4 shows the result of an analysis using a method based on a modulated input signal for compression ratio 4. The effectiveness of the compressor is determined by measuring the difference between the spectral levels of carrier and first sideband. Figure 4 shows the effect of the delay for the optimal condition. The effective compression ratio at a modulation frequency is given for the optimum delay of 3 ms (circles), for no delay (triangles pointing downward), and for a delay of 10 ms (triangles pointing upward). It shows that a well chosen delay makes the compressor effective for frequencies up to about 20 Hz. Switching the delay off reduces the effectiveness considerably.

Compensation for the altered dynamics of hearing (recruitment) should thus not be based on the static characteristics but should take full account of the limitations of the temporal and spectral resolution of the (impaired) ear. The limitations can even be used to hide from detection the distortion caused by the signal processing.

Conclusions

1. New technology should be checked separately for normal hearing people and for the hearing impaired.

2. Presentation of speech cues above threshold should be distinguished from presentation of detectable cues. Poor spectral and temporal resolution may make suprathreshold cues undetectable.

3. Technology in hearing aids should take into account the limitations of impaired hearing as described by the spectral and temporal resolution; it should even try to use these effects to prevent distortion products from being detected.

4. Compressors in commercial hearing aids should be made much faster if they should work as syllabic compressors. Overshoot suppression should be added.

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