Hearing with Hearing Aids in Noise and Reverberation

Hans Verschuure and Lau Nijs

Introduction

Patients complain often about problems in communication, even with properly fitted hearing aids. These difficulties are often experienced in situations with a lot of noise or in reverberation. It seems naturally in cases of complaints to check the correctness of the fitting by real-ear measurements and to verify the expectations of the patient with regard to the hearing aid to exclude unrealistic expectations. Nevertheless complaints remain common. The situations that are described here as noisy and reverberant can be the cause of the complaint and not an improper fit. This paper deals with an approach to see what the problems are and what solutions are available.

Goal of Rehabilitation

First of all we should realize that hearing-aid fitting is part of a rehabilitative effort, which may include further actions. In the new International Classification of Functions published by WHO in 2001 (WHO, 2001) a problem in a health conditions can lead to an impairment of a function or structure. An impairment will lead to limitations in the activities of the sufferer, which will restrict his or her participation in society. The major change between the old classification (WHO, 1980) and the new one is that also environmental factors have to be taken into account. This means that limitation in the activity can be caused by the impairment, but the circumstances in which the impaired person has to function are also relevant. Rehabilitation in this context means that a person has to be adapted maximally to the environment and the environment has to be adapted to a person with special needs. An advantage of this approach can be an improved and easier communication for a non-impaired person in the same disadvantageous conditions. In this context, the solution to hearing problems in reverberant conditions should also be sought in e.g. reducing reverberation in rooms where aural communication takes place.

Problematic Situations

The problem described in the introduction is the limitation of communication by background noises and reverberation. We came across this problem when we started seeing persons with a mental handicap. Studies show a much higher prevalence of hearing loss in this group because of early ageing in patients suffering from e.g. the Down syndrome or by a common cause for the mental problem and the hearing loss like oxygenation problems at birth (e.g. Mul et al, 1997; Evenhuis, 1995; Evenhuis et al, 1992; Cooke, 1988; Buchanan, 1990).

People with mental disabilities often live and work under more unfavorable acoustical conditions than normal-hearing persons do. In our country there is a tendency not to institutionalize them but to have them live in normal community housing, under supervision. To minimize the cost of supervision it is not unusual to house them in two houses from which the common wall is removed. In such a home six to eight people live with supervision. Carpeting and decoration of these enlarged living rooms is often rather simple because of hygienic reasons. This makes the living room very reverberant. The acoustical conditions in their working place are often similar. Supervision requires good sight of a supervisor on...
the working place of many people and thus they work in large and open halls.

We all know from experience that hearing-impaired persons perform poorly in those large rooms and halls (e.g. Plomp and Mimpen, 1979; Duquesnoy and Plomp, 1980). We therefore asked ourselves whether the provision of hearing aids under those conditions was realistic or that some further rehabilitative measures should be taken with regard to room acoustics.

So far we were concerned with people with a learning disability. However, living conditions can be rather similar in modern-design houses. Design decoration is often limited with wooden or tiled floors and quite large rooms. We see that also for people with normal mental capacities disadvantageous conditions can occur and thus can be the cause of communication problems with hearing loss. Clinical audiologists should therefore have a basic knowledge of room acoustics and ways to improve room acoustics.

In this paper we deal with the following questions:

- Can we measure the needs of individual patients for hearing in a background noise and with reverberation?
- Can we assess the solutions in those conditions where a hearing aid does not give sufficient help?
- Can we differentiate between possible solutions like provision of assistive devices and improvement of room acoustics?

Principles of Room Acoustics

A sound produced in a room is reflected by all walls including floor and ceiling. The reflections cause the sound to remain in the room and to die away slowly. This effect is called reverberation. The principles of reverberation are described by a small number of principles and parameters. The most important ones are:

1. Absorption. It is a quantitative measure of the amount of sound energy that is absorbed by the wall at each reflection. The absorption coefficient of a material is defined as the fraction of the sound energy that is not reflected from the wall. It means that an absorption coefficient of 0.3 means that 70% of the energy is reflected back into the room, 30% is either absorbed by the material or transferred into some other material.

2. Reverberation time. It is a measure of how a room reacts to a sound. It is often defined as the time required for a signal to reach a level 60 dB down from the original level. It is determined by the time required for the signal to travel from one reflection to the other (volume of the room) and by the amount of absorption that takes place at each reflection.

3. Direct sound. This is the sound that travels directly from the generator to the listener (or from the loudspeaker to the microphone) without being reflected by the sides of the room or objects in the room. The level of the direct sounds decreases by 6 dB per doubling of the distance.

4. Near field and far field. Above a certain distance from the sound generator, the level of the sound is no longer determined by the direct sound. All the reflected waves produce a higher level than the direct sound. We differentiate between the near field where the sound level is determined by the direct sound and the far field where the sound level is determined by all the reflected sound.

5. The reverberation radius. It is the distance from the sound generator (speaker) where the sound level of the far-field sound equals the sound level of the near-field or direct sound.

In figure 1 we present the sound level as a function of the (normalized) distance. For a distance shorter than 1 we are in the near field. For the distance between 1 and 4 there is still some effect of the direct sound, so often being in the far field is defined as being at a distance of more than 4 times the reverberant radius.

![Figure 1. Relationship between level of sound and the distance from source, normalized for reverberation radius.](image-url)
Reverberation is present in all rooms. If more absorption is added to a room there is no effect on the direct sound, but the reflected sounds die faster and a shorter reverberation time can be achieved.

**Theory of Reverberation and Speech Intelligibility**

**Reverberation Time**

The reverberation time in a certain room is given by a simple equation described by Sabine as

\[ T_{60} = \frac{V}{6A} \]

with

- \( V \) is the total volume of the room
- \( A \) is the amount of effective absorbing surface, often described as

\[ A = a_1 S_1 + a_2 S_2 + a_3 S_3 + \ldots \]

with

- \( a \) the absorption coefficient
- \( S \) the total surface with this absorption coefficient

From the equation we see that doubling of the room volume means doubling of the reverberation time and that doubling of the effective absorption surface halves the reverberation time.

**Reverberation Radius**

The equation for the reverberation radius is

\[ r_{rev} = \frac{1}{4} \sqrt{\frac{QS}{\pi S - A}} \]

with

- \( Q \) being the directional factor of the source
- \( S \) being the total surface
- \( A \) being the equivalent surface of effective absorption as in the equation for the reverberation time

The equation shows that the addition of absorbing material will lengthen the reverberation radius. In a normal room the reverberation radius is usually between 1 and 1.5 meters.

**Reverberation Time, Intelligibility and Sound Quality**

Reverberation has two effects on speech intelligibility, an effect on the level and an effect on temporal structure of the sound. Early reflections contribute to the level of the sound without affecting the temporal structure much and thus increase intelligibility of the speech. This applies to about the first 60 to 80 ms of the sound. Later arriving sounds interfere with parts of the speech following this particular sound and thus have a masking effect on the following speech. In short, early reflections contribute to intelligibility, late reflections deteriorate intelligibility.

Added absorption has no effect on the direct sound. The early reflections are reduced somewhat by the added absorption but later reflections are reduced considerably. It will result in a shorter reverberation time and a longer reverberation radius, as can be seen from the equations. As such added absorption will improve intelligibility as long as the level of the speech does not get too low. In figure 2 (above) we show a speech recording of a sentence recorded at one and at five meters, in a non reverberant room with a reverberation time of less than 0.1s and in a reverberant room with a reverberation time of 10 s.

We see that the speech in the non-reverberant room at one and five meters looks very similar, except for a level reduction, indicating the effect of the direct field. In the reverberant room we see that all sounds...
are smeared out with almost no level difference between one and five meters; both recordings are in the far field. It is quite clear that the speech features are well preserved in the non-reverberant room but are distorted in the reverberant room.

This suggests we should aim at the shortest possible reverberation time in order to optimize speech intelligibility. A short reverberation time is better for speech intelligibility in general except for a limitation in level. However, sound quality deteriorates with short reverberation times. The room gets dry acoustics and music sounds awful.

To summarize direct sound has good intelligibility, early reflections contribute to level and have no negative effect on intelligibility, late reflections contribute to level, can improve perceived sound quality but may result in poor speech intelligibility.

Theory of Speech Intelligibility

At this point we need a method to describe speech intelligibility. For calculations the Speech Intelligibility Index (SII) is often used (ANSI, 2002). Unfortunately, this takes into account only effects of elevated threshold and masking noise, not the effect of reverberation or time modulated noises. Steeneken and Houtgast (1980) developed a method called the Speech Transmission Index in which also temporal effects are taken into account and thus also reverberation.

The principle is on first view similar to the approach of the SII. Contributions from the various frequency bands contribute to speech intelligibility with different weighting factors for the different frequency bands. Only information above threshold and above masking level contributes to intelligibility and only a 30-dB range is relevant.

The difference between the methods is that for the STI the contributions in each frequency band are determined not by level differences but by the amount of preserved modulation in each frequency band. For the theory we should analyze the signal present in a certain (octave or third-octave) frequency band. For this signal we analyze the amount of modulation that is present. To do this the modulation spectrum of each frequency band is determined. Relevant modulations range between about 0.1 and 32 Hz. In a similar approach as for the spectral frequency bands, the modulation frequencies are given weighting factors and the contribution per spectral frequency band (0.25 to 8 kHz) is determined by summing the preserved information content per modulation frequency band (0.1 to 32 Hz). For more details, please, see the original publication or later publications.

In summary we see that the STI takes into account the effect of spectral masking in a similar way as the SII does but adds to it the effect of modulation masking, thus allowing for reverberation to be taken into account.

The two effects of masking noise and reverberation are represented by two separate parts of the equation for the modulation transfer function:

\[
Reverberation:\ m(F) = \frac{1}{\sqrt{1 + \left(\frac{2\pi FT}{13.8}\right)^2}}
\]

\[
F \quad \text{the modulation frequency}
\]

\[
T \quad \text{the reverberation time}
\]

And

\[
Masking:\ m = \frac{1}{1 + 10^{((-S/N)/10)}}\]

\[
S/N \quad \text{the signal to noise ratio}
\]

With these equations the effects on the STI can be calculated of masking and reverberation. For tests under headphones, we have no reverberation. Tests show that for simple sentences the required signal-to-noise ratio for a normal-hearing person to understand 50% of the speech is about −4 dB (Plomp and Mimpen, 1979; Verschuure and van Benthem, 1992). There is a simple relationship between this critical signal-to-noise ratio and the STI:

\[
STI = \frac{(s/n) + 15}{30}
\]

for \(s/n\) between −15 and +15 dB

This means that for understanding of these sentences by a normal-hearing person a STI is required of 0.33.

Requirements of Patients

Hearing impairment causes a reduction in the spectrum analyzing capacity of the ear. A consequence is that the speech signal and the background noise can be separated less effectively by a hearing-impaired person. This means that the possibilities of the hearing impaired person to understand speech are poorer. Verschuure and van Benthem (1992) have shown that the change in critical speech-to-noise ratio is not determined by the amount of loss,
they found no correlation with the pure-tone average over 0.5, 1 and 2 kHz; they found a better correlation with the high-frequency hearing loss, averaged over 2 and 4 kHz.

Critical Signal-to-noise Ratio

In figure 3 we show some of their results. The figure shows that the s/n ratios differ considerably between patients. Some have a 50 dB loss (pure-tone average over 2 and 4 kHz) and an almost normal critical signal-to-noise ratio. Others with a loss of 50 dB have a critical s/n ratio of about +8 dB, a deterioration of 12 dB. If we take into account that each 6 dB means halving of the distance at which one can understand speech at a party, the change means that the normal conversation distance of about 1 m has to be reduced to 25 cm, a socially unacceptable distance.

Next we see that, given a high-frequency hearing loss of up to about 60 dB, we expect a person to be able to communicate only if the required s/n ratio is better than about +6 dB. Some will still not function in those conditions, but most of them will. For the remainder we take this value as our target value to make speech communication with hearing-impaired people possible.

STI, Signal-to-noise Ratio and Reverberation Time

Duquesnoy and Plomp (1980) tested whether the STI theory holds when we compare in elderly people the effects of a steady background noise with those of reverberation. Healthy elderly persons with different degrees of hearing loss were tested under different reverberant conditions and their results were compared with those of a reference group of normal-hearing listeners. The effect of reverberation was added to the signal-to-noise ratios by calculating the STI for different conditions. The results show a constant minimum STI value for different reverberant conditions indicating a constant minimum amount of information over different reverberant conditions.

In figure 4 we show the STI values for different reverberation times and signal-to-noise ratios. The dotted lines represent lines of equal signal-to-noise ratio.

The figure shows clearly the STI drops with a longer reverberation time.

For patients we can determine the minimum signal-to-noise ratio they require for understanding speech by measuring this value of the s/n ratio with headphones. Because there is no reverberation, this value can easily be transferred in STI values, e.g. a s/n ratio of +6 dB means a required STI value for understanding speech of 0.7.

Figure 4 shows something that is usually not realized by clinicians. If we take our example again of a person with a critical s/n ratio of +6 dB, we see that a STI value of 0.7 is required for understanding.
speech. It also means that such a person, when in the far sound field, cannot understand speech in a room with a reverberation time of more than 0.5 s. The implications are clear:

- It is no use talking to this person from a distance, not even in most living rooms; it is out of the question in an office even when acoustics are considered pretty good by normal-hearing persons.
- A weak disturbing sound makes communication impossible if the reverberation is just short enough to allow speech communication.
- Fitting of a hearing aid for a person living or working under those conditions will not solve the problems in communication. This has nothing to do with a poor fit, nor with a poor quality of the hearing aid. It is simply caused by the required amount of speech information. Because of the importance of the higher frequencies for understanding in noise, good attention should be paid to the high-frequency gain.
- There is a need for acoustic measures in an environment with many hearing-impaired persons, even if they suffer from mild presbycusis. Based on this finding we have proposed in our country to require a STI of 0.7 in the living quarters of people with a learning disability. Similar values would be required in homes for the elderly. It poses big problems to acousticians, which can be dealt with efficiently only in the design stage.
- In many cases assistive devices are needed for communication like directional microphones, loop systems, infrared systems or FM systems.

**STI Values and Room Acoustics**

The definition of the STI allows for a measurement of the STI for a given room and positions of speaker and listener (Steeneken and Houtgast, 1980). There are devices on the market to measure these values and the STI can be calculated in a model. In a research project two excel sheets were designed:

1. A model to calculate the STI values from the dimensions of a room, the acoustical properties of everything in the room and the distance between speaker and listener. The model is only valid if not too close to walls (Dingemanse, unpublished).
2. A sheet with the acoustical properties of all building material available at present to make choices between material possible and to see the effect of price (van Berloo, unpublished).

A comparison between measured values and calculated values for a small number of rooms showed that the model was satisfactory.

We calculated the STI values in a typical room where people with learning disabilities lived and found that communication for two people sitting on the couch was possible (STI = 0.72) but not with the television set switched on. When a supervisor would enter through the door from the office, (s)he could not be understood (STI of 0.53). It is obvious that this can easily result in behavior described as stubborn, uncontrollable and similar in people with learning disabilities, while it is only a matter of speech intelligibility. Similar conclusions may be drawn for older people.

**Hearing Aids and Noise Reduction**

**Noise-reduction and Annoyance-reduction Algorithms**

The problem of communication in noisy environments is often recognized and there is a provision for noise reduction in many modern hearing aids. Unfortunately most of these noise reduction schemes reduce the gain when the input signal is recognized as having no speech characteristics based on certain criteria. In most cases such a system will not help to understand speech better in a noisy situation because it reduces the level of the speech and the level of the noise or reverberation by the same amount and thus not changing the speech-in-noise ratio. It may reduce the annoyance of people speaking loud in noisy circumstances. I would prefer to call these systems annoyance-reducing systems.

A real noise reduction means that a speech signal is amplified more than the background noise. In cases of two speakers this is practically impossible for a hearing aid because the hearing aid cannot differentiate between the wanted speaker and the jamming speaker. Algorithms have been developed to improve signal-to-noise ratios for speech in a noisy non-speech background, but mostly the deterioration of speech intelligibility from distortions is larger than the gain in signal-to-noise ratio. It can be expected that binaural processing may provide some gain in the future (Peissig and Kollmeier, 1997). Further research is needed.
Directional Microphones

Workable solutions have been introduced requiring a spatial separation of the sources. Now a directional microphone can reduce the level of the background noise assuming it the jamming sound does not come from straight forward where the speaker is. The user has to face the speaker directly and cannot turn the head.

There have been a number of such aids introduced:

1. Directional microphone. In general we can say that a single directional microphone has a very limited effect on the signal-to-noise ratio. The improvement is usually not more than 1 dB. ITE solutions perform better than BTE solutions.
2. Small array microphones using two or three microphones on a hearing aid. The effectiveness of array microphones depends very much on the distance between the microphones, because of the direct link with the wavelength of the sound. As such the systems are rather ineffective in ITE hearing aids and are much more effective in BTE hearing aids. The effects in a diffuse sound field are usually limited to a noise reduction by about 4 to 5 dB. However these 4 to 5 dB may be very relevant to a patient. Effects in cases of isolated jamming speakers in the near field may be much larger and favorable results are often obtained under these experimental conditions.
3. Array microphones placed on an external device used as input to a hearing aid. External array microphones have been marketed or are under development. Studies by Soede et al. (1993) and Merks (2000) in our clinic have shown an effect of about 8 dB for an array length of about 10 centimeters.
4. Single, directional or array microphone placed near the speaker; the signal is transferred to the user by induction, infrared or wireless. In those cases the microphone is placed in the near field at a distance of a few centimeters from the mouth of the talker, usually much further away from jamming sounds. There effectiveness can be very high.

Some remarks have to be added in reverberant conditions. Directional microphones in general are only effective if they can pick up the direct sound; it can reduce the reverberant sound field that has an omni-directional incidence of sound. It means that they are only effective if there is a relevant direct field. This is usually limited to the direct field and will certainly not exceed a distance of about 3 to 4 times the reverberant radius. If the speaker is further away from the listener, directional microphones have little effect. A reduction of the signal-to-noise level by an array microphone by 6 dB would mean that the reverberant radius is doubled and that effects of noise reduction can be experienced at a double distance from the speaker, but not further away.

In the case the listener is in the far field, also the far field of loudspeakers when used, directional microphones do not improve the signal-to-noise ratio and do not contribute to a better intelligibility. Placing a microphone in the direct field and transferring this signal to the listener is the only way to achieve a better intelligibility in such conditions.

Conclusions

1. The needs of a patient with a hearing problem in a noisy environment and in reverberation can be assessed by the measurement of the critical speech-in-noise threshold and transferring this number in a critical STI threshold.
2. The STI value in relevant rooms for the patient can be measured or estimated. From the comparison of the STI value in the room and the critical STI value of the patient the required improvement in signal-to-noise ratio can be determined.
3. We can compare the effects of possibilities to improve the STI value with the required need and thus make a choice of a well-tailored device in a given condition.
4. The effects of changing the acoustic properties of the room can also be investigated and taken into account. Cost-effectiveness may be an important issue.
5. Clinicians should be able to assess these effects or have them assessed in order to help the impaired listener experience less limitation in activity and participate better in society.

References
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