ARCHIVES OF ACOUSTICS
29, 3, 397–409 (2004)

DIGITALLY CONTROLLED ACTIVE NOISE REDUCTION WITH INTEGRATED SPEECH COMMUNICATION

H.J.M. STEENEKEN, J. VERHAVE

TNO Human Factors, Soesterberg, The Netherlands
email: herman@steeneken.com, verhave@tm.tno.nl

(*received 2 April 2002; accepted 30 January 2004*)

Active noise reduction is a successful addition to passive ear-defenders for improvement of the sound attenuation at low frequencies. Design and assessment methods are discussed, focused on subjective and objective attenuation measurements, stability, and high noise level applications. Active noise reduction systems are suitable for integration with an intercom. For this purpose the intelligibility in combination with environmental noise is evaluated. Development of a system includes the acoustical design, the (digital) feedback control, and the speech input facility. In order to achieve optimal performance a specific audio design is required. An example of such a development is discussed.

Key words: hearing protection, active noise reduction, speech communication, digital controlled feed back.

1. Introduction

Active noise reduction (ANR) is an effective tool to increase the sound attenuation of hearing protectors. Especially for the low frequency range the passive sound attenuation of an earmuff or earplug is often insufficient. ANR can provide an additional attenuation of 20–30 dB at low frequencies (system dependent approximately 500–1000 Hz).

In an earlier study two analogue active noise reduction systems have been developed, one system based on an earmuff and a second system based on an earplug. The earmuff-based system offers a high additional attenuation (up to 25 dB) and is used at very high noise levels (up to 160 dB SPL).The earplug-based system is small and applicable in combination with a gasmask or with a pilot helmet. Recent developments are based on a digitally controlled system and also allow for monitoring the noise dose.

A specially developed speech interface is used for injection of speech signals from an intercom system. Advanced signal pre-processing results in high quality speech and low acoustic distortion. Assessment methods of ANR systems differ from methods as used for passive hearing protection. Due to the non-linear characteristics no measurements at the threshold of hearing can be performed. Various objective and subjective assessment methods of the attenuation and the speech quality will be discussed.

2. Principle of active noise reduction

Active noise reduction is based on the addition of a secondary sound signal to a primary sound signal which has to be suppressed (LUEG [1]). If the waveform of the two signals are identical but in anti-phase the resulting sound will be zero. A perfect match is theoretical; in practice a feedback loop is used according to the block diagram given in Fig. 1.

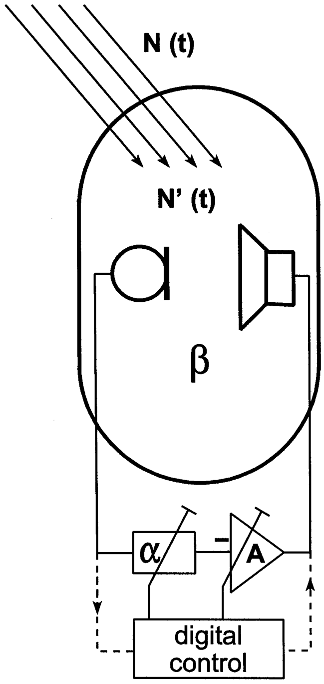

Fig. 1. Schematic diagram of an active noise reduction system within the shell of a hearing protector.

The resulting noise signal $N'(t)$ at the microphone position is the sum of the primary noise signal $N(t)$ (leaking from the outside of the hearing protector) and the secondary compensation signal from the loudspeaker. The latter signal is equal to the resulting noise signal at the microphone multiplied by the loop gain, hence:

$$N'(t) = N(t) - N'(t) \cdot \alpha \cdot \beta \cdot A, \tag{1}$$

$$N'(t) = \frac{N(t)}{1 + \alpha \cdot \beta \cdot A}, \tag{2}$$

where β represents the frequency transfer and the efficiency of the electro acoustic transducers (microphone, and loudspeaker), α represents the frequency transfer of the correction amplifier and A the gain of the output amplifier. The amount of suppression is given by the denominator of Eq. (2). An increase of the loop-gain ($\alpha \cdot \beta \cdot A$) results in more suppression.

The frequency transfer of the combination of the electro-acoustic transducers and the cavity under the earmuff is limited. An example of the frequency transfer function (amplitude and phase response) is given in Fig. 2. In view of this transfer function three ranges for the denominator of Eq. (2) are identified:

1) the denominator is greater than one which results in a suppression of the primary noise,
2) the denominator is smaller than one but greater than zero which results in an amplification of the primary noise, and
3) the denominator becomes zero which results in an unstable system which will oscillate.

The last two possibilities should be avoided by either a lower loop gain or correction of the amplitude and phase response. Such a correction can be obtained by a compensation network to be included in filter α or reduction of the gain of amplifier A. Reduction of the total loop-gain results in a smaller amount of noise suppression. Therefore, a careful design of the acoustical properties of the cavity within the earmuff, and careful selection of the transducers with an optimal frequency response is required. The frequency and phase response of the compensation network is defined in relation to the frequency response given by β and amplifier A. The loudspeaker (or telephone cartridge) is the most critical part of the loop. In general a system with a good response – at very low frequencies is required. A description of the design criteria is given by OLSON and MAY [3] and NELSON and ELLIOTT [2] and STEENEKEN and VERHAVE [5, 6].

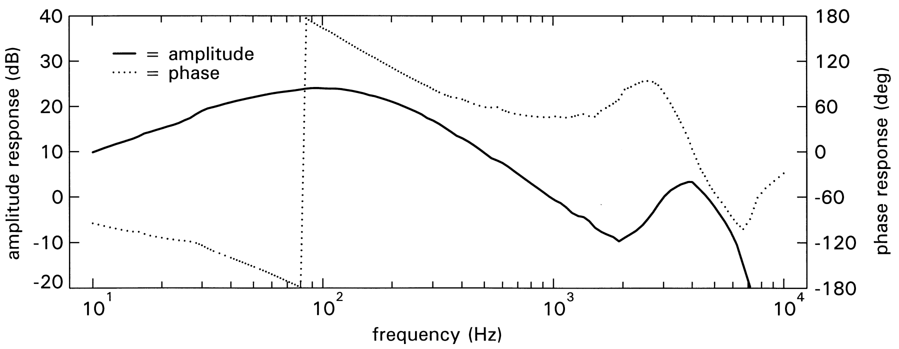

Fig. 2. Amplitude and phase response of the combination of a telephone and microphone within an earmuff placed on the head of a subject.

A better representation of the stability issues related to the frequency response is offered by a Nyquist diagram which is the vector representation of the loop gain ($\alpha \cdot \beta \cdot A$). This is represented in Fig. 3. No part of the curve should include the instability point (-1, dot in Fig. 3) at which the loop gain is equal or lower than "-1" and the phase response is $180°$ degrees instead of $0°$. Two parameters have to be optimised for maximum loop gain outside the restricted area. These are the gain of amplifier A and a frequency dependent correction by filter α. For analogue systems these parameter settings are tuned once at a stable position (normally 6 dB below instability). This is done to make the system performance user independent. However, this tuning is not optimal as individual users may have a different effect on the frequency response especially in the earmuff cavity.

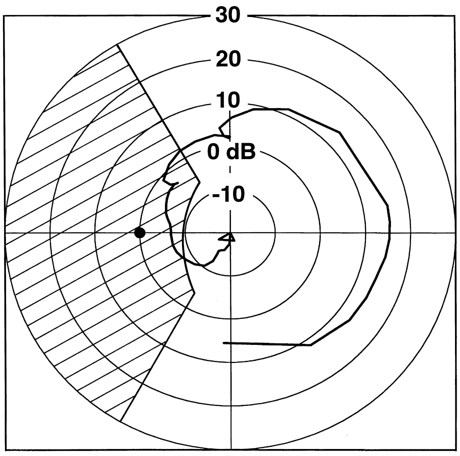

Fig. 3. Nyquist diagram according to the frequency response of Fig. 2.

We developed a method to determine the stability of the feedback loop dynamically. By inserting an impulse at the telephone side (see dotted line of Fig. 1) the impulse response of the closed loop can be obtained at the microphone output. An example of this impulse response is given in Fig. 4. The slope of the decay is a measure for the stability. This slope should be below a certain value (see also Fig. 5, > -1 db/ms, the maximum response can be adjusted to 20 db/ms and this allows to adapt to non-stationary noise). The digital controller will now dynamically adjust the amplifier gain (A) in order to optimise the loop gain. Also the frequency response of the correction filter can be adjusted with this method.

The impulse response can be determined continuously. For this purpose it is required that the user is not disturbed by the impulses injected in the loop. Therefore, the impulse level is very low. In order to obtain a useful response a number of impulses are added, hence the level of the added (correlated) responses increases equal to the number of additions while the uncorrelated background noise increases with the square root of the number of additions. Hence, an improvement of the impulse-to-noise ratio is obtained and a dynamic control of the system parameters is achieved.

When the controller samples the microphone output, a simple algorithm could be added to calculate the noise dose from the same signal. For this purpose a standard frequency weighting according to the A-curve is applied. The intensity is integrated and converted to an equivalent intensity for an exposure of 8 hours (the noise dose).

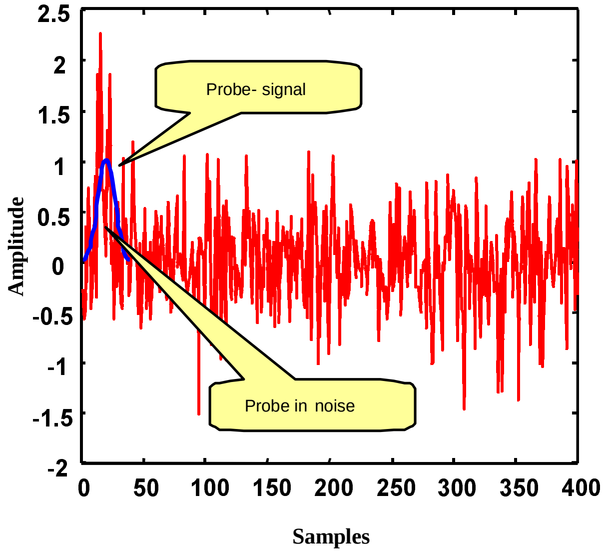

Fig. 4. Impulse as injected in the system loop. Notice the background noise level.

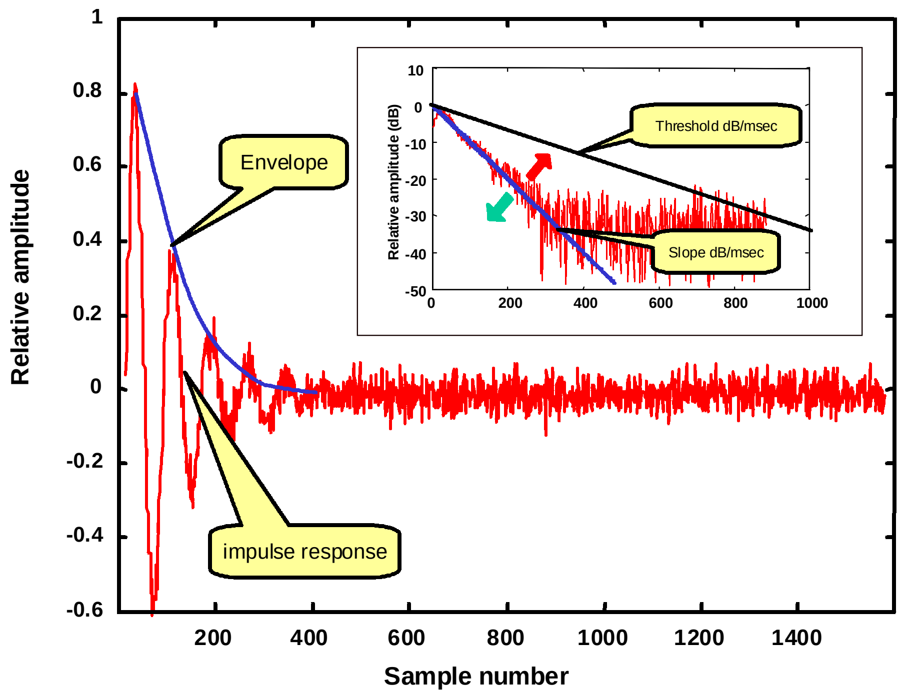

Fig. 5. Sum of many added impulse responses of the injected impulses. The decay of this response gives a measure for the stability.

3. Speech communication

If speech communication is not properly processed, it will also become suppressed by the ANR system.

An optimal method to prevent this, is to compensate for the feed back on the speech signal. Compensation can be done with the circuit given in Fig. 6. As the feed back loop will also suppress the speech signal, the loop has to be (electrically) opened. This is done by subtracting the speech signal (from the microphone signal) at the beginning of the loop. This is theoretically correct if the transfer from telephone to microphone has an ideal frequency response. However, it is not the case due to the telephone and cavity response. Therefore the signal to be subtracted is processed by a FIR filter (F1) which has a similar response as the transfer within the earmuff. The digital control systems determines this response automatically and converts this response to a FIR filter characteristic. As F1 will give a delay typical for FIR filter a similar delay has to be introduced in the primary speech channel. This is accomplished by F2 which has a flat response and a similar delay as F1.

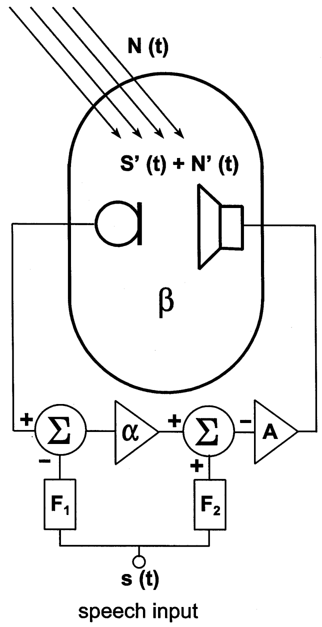

Fig. 6. Schematic diagram of an active noise reduction system including the addition of speech signals.

4. Assessment of ANR systems

The performance of an ANR system depends on a number of technical properties. Of course the addition of active sound attenuation is a major aspect. However, in order to specify the *personal* protection and safety, and not just mean values, the following items are of interest:

- passive sound attenuation as a function of frequency,
- active sound attenuation as a function of frequency,
- variance among systems,
- variance among users,
- stability on the head,
- stability for open system during placing or removing from the head,
- sensitivity for vibrations,
- maximum sound pressure level (dynamic range),
- overload response,
- speech intelligibility of the integrated communication system.

In this overview we will focus on the sound attenuation and speech intelligibility. It should be mentioned that an international Round Robin is in progress in order to determine optimum assessment methods specifically related to the adverse Military environmental noise conditions.

4.1. Sound attenuation

With the introduction of active hearing protectors, which may introduce some system noise at the users ear, the assessment of the sound attenuation according to the standard measuring methods (ISO4869-1) is no longer valid. The ISO method is based on the threshold of perception and, thus, limited to low sound levels. The noise introduced by the ANR systems will interfere with the measurements. Also the sound attenuation of ANR systems is level dependent due to possible overload. Hence, measurements should be performed at various levels.

Three alternative methods for measuring the sound attenuation are in use:

1. By comparing the sound pressure level measured under the earmuff with the ANR system switched on and off. The level difference between the two measurements gives the sound attenuation. The measurements are performed by making use of the sense-microphone included in the ANR-loop.
2. Similar measurements as described under 1 by making use of an additional microphone, positioned close to the entrance of the ear canal.
3. By subjective matching of the loudness of two sound levels, representative for the additional attenuation of the ANR system. (Zera [7] proposed a method to match masking as a function of frequency).

4.1.1. Objective measurements

The active sound attenuation can be obtained by measuring the difference between the sound pressure levels under the earmuff shell with the ANR system switched on and off. As measuring microphone the loop microphone or an additional microphone placed near the entrance of the ear canal can be used. By means of a positioning system the miniature microphone is placed near the entrance of the ear canal (Fig. 7). This method is called MIRE (Microphone In Real Ear) and is considered to become a new international standard (Technical Committee CEN/TC 159).

Fig. 7. Measuring microphone near the entrance of the hearing canal.

Preferably, the noise level and spectrum used for the measurements are identical to the noise level and spectrum of the real application. As ANR systems may have a level dependent attenuation it is advised to determine the attenuation as a function of the noise level.

The attenuation is measured as a function of the frequency. Usually a resolution of 1/3 octave band is used. For this purpose the output signal of the microphone used for the measurements is analysed by a spectrum analyser.

In order to obtain representative results and to get information on user dependency, various subjects are used.

4.1.2. Subjective measurements

For the subjective assessment of the attenuation of active hearing protectors a subject (with an ANR system for each ear) is placed in a diffuse sound field which alternates periodically between two levels (typically every second). An example of this level alternation is given in Fig. 8.

During the highest sound pressure level the ANR system is switched on, while during the low sound level the ANR system is switched off. The subject will hear a smaller difference between the two sound levels as the ANR system attenuates only the highest

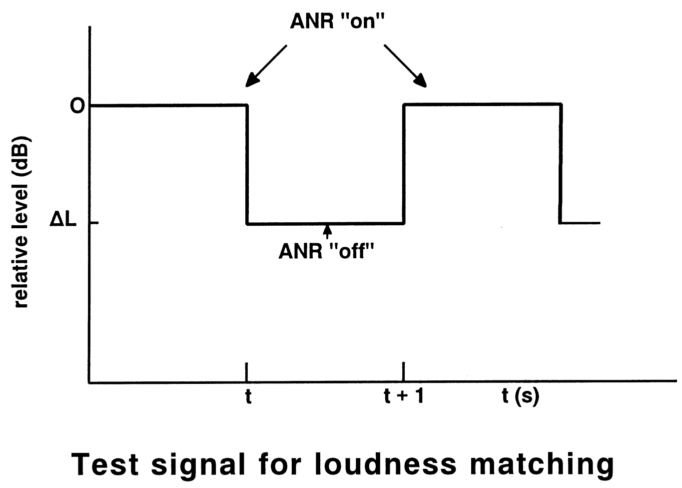

Fig. 8. Relative test signal level as a function of time for the subjective measurements of the suppression of an ANR system. The ANR system is switched on and off simultaneously with the test signal envelope.

level. The subject is asked to match both levels for equal loudness by adjusting the level difference ΔL between the two signals. The resulting difference in sound level outside the earmuff is equal to the subjective attenuation provided by the ANR system. The adjustment can be made by changing the sound level during the "ANR-off" interval. Since the subject adjusts for a continuous signal, the on/off rhythm is indicated with a light signal. A study showed that the accuracy lies within 1–3 dB.

The measurements have to be performed in a specific room with a diffuse sound field. The test signals that are used consist of noise bands with a bandwidth of 1/3 octave. Measurements are performed in one-octave steps. The absolute signal level can be adjusted to any level, which is high enough not to interfere with the system noise. However, as the noise reduction of ANR systems may be level dependent, the measurements should be performed systematically as a function of the level.

4.1.3. Comparison of subjective and objective measuring results

A comparison between subjective and objective attenuation measurements was made. The subjective attenuation was measured with four subjects and various signal levels. For one of the conditions the 1/3 octave band signal level was 110 dB SPL. The mean attenuation for these conditions, as a function of frequency with one octave steps, is given in Fig. 9.

The objective attenuation was measured with the loop microphone as well as with a special electret microphone positioned close to the entrance of the ear canal. For the objective measurement a pink noise (level 105 dB SPL) was used. The results indicate that the attenuation values obtained with the subjective method and those obtained with the ear microphone (MIRE) are in close agreement. The attenuation values obtained

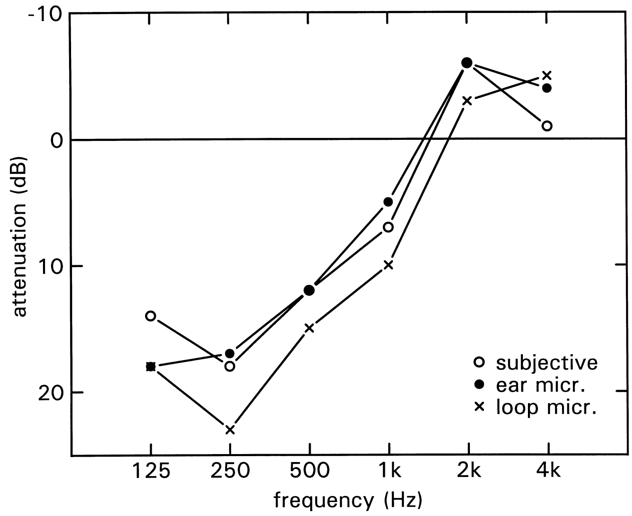

Fig. 9. Mean sound attenuation measured with 4 subjects in one octave intervals for the subjective and objective methods.

with the loop microphone are somewhat higher (2–5 dB). Obviously, the sound field under the earmuff is not homogeneous and is minimal at the sensing position of the loop microphone.

An example of some results of a study on passive and active sound attenuation of an in house system is given in Fig. 10. The graphs represents the mean attenuation and the standard deviation for three identical systems and 4 subjects.

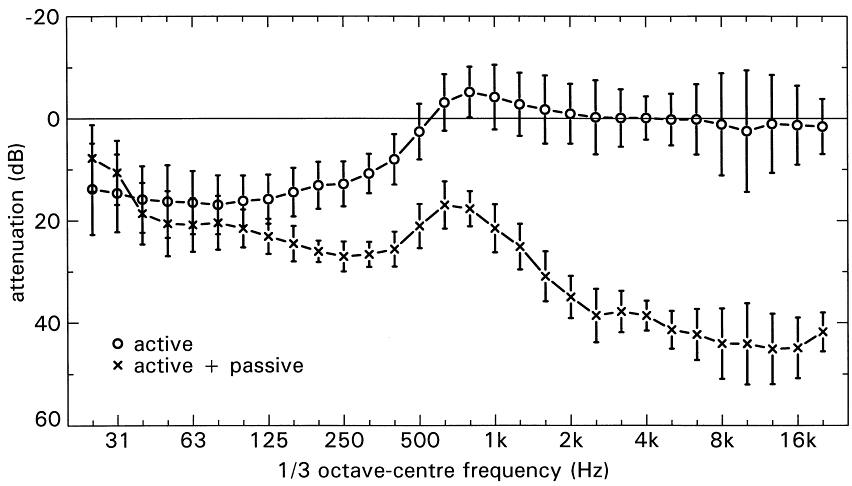

Fig. 10. Mean active attenuation and standard deviation for 3 systems and 4 subjects (8 ears). The total attenuation (passive and active) is also given.

4.2. Speech transmission quality

The speech quality depends on the method used for the injection of the speech signal. Some systems make use of the method given in Fig. 6 while others inject the speech signal at the sense microphone input. Some designs make use of a correction amplifier.

As the speech transmission quality is defined by the design of the ANR system, the speech injection method, and the suppression of background noise, it is important to assess the speech intelligibility in a representative condition.

This assessment can be done with subjective measures (by making use of speakers and listeners) or by objective methods (by making use of a measuring device). In this study an objective method (the Speech Transmission Index, STI) is used (STEENEKEN and HOUTGAST [4]; IEC 60268-16).

The STI-method assumes that the intelligibility of a transmitted speech signal is related to the preservation of the original spectral differences between speech sounds. These spectral differences may be reduced by band-pass limiting, masking noise, temporal distortion (echoes, reverberation, and automatic gain control), and nonlinear distortion (system overload, quantization noise). The reduction of these spectral differences can be quantified by the *effective* signal-to-noise ratio obtained for a number of frequency bands. Also human related hearing aspects such as masking, the reception threshold, hearing disorders, and non-native speakers and listeners may reduce the effective signal-to-noise ratio. The method is based on the calculation of the effective signal-to-noise ratio in seven relevant frequency bands (octave-bands, center frequencies ranging from 125 Hz to 8 kHz). Weighted contributions of the quantified information transfer function in seven octave bands results in a single index, the STI_r.

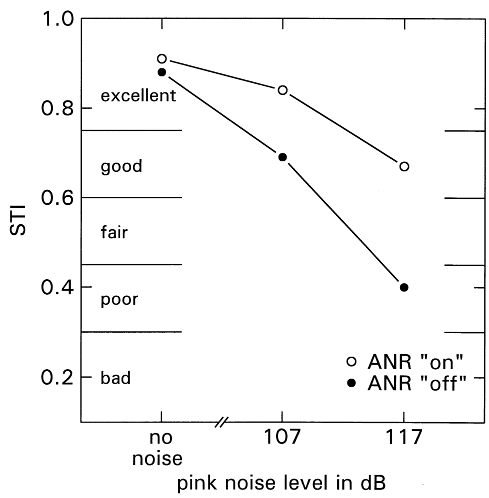

Fig. 11. STI at three noise levels for an ANR system switched on and off. As measuring microphone under the ear-shell the MIRE microphone was used.

The STI is obtained by applying a specific speech-like test signal at the audio input and by analysis of this transmitted test signal through the same measuring microphone as used with the MIRE attenuation measurements. The STI for a specific communication system with ANR as a function of the noise level is given in Fig. 11. The STI is given for two conditions: ANR switched on and off.

Hence, the effect of the ANR on the STI-value can be obtained by comparing the two conditions. Additional to the STI-value also a qualification (based on STI) is given. The improvement of the speech transmission quality is obvious. It is shown that for a constant speech intelligibility (STI $= 0.7$) a 10 dB higher noise level can be applied. Hence the *effective* gain in this situation and for this type of noise is 10 dB.

5. Comparitative results

A comparison of some commercial systems was made. We investigated both the passive and the active attenuation. It was found that the stability of some systems was such that the system started to oscillate when placed on the head of a subject.

Two types of oscillation were found (1) a very low frequent oscillation (below 5 Hz) or above 1000 Hz. The sound pressure levels during these instabilities were very high. For this reason we adjust our system 6 dB below the point of instability. If the performance of systems is compared, this security range is often not included. One might get an impression of the stability by observing the amount of negative attenuation. A typical value is 6 dB around 800–1200 Hz. Some systems show a value of over 12 dB. These system are generally not stable. In Fig. 12 a comparison is given for 5 commercial ANR systems (labelled C-G) and two version of the system discussed above (labelled A and B).

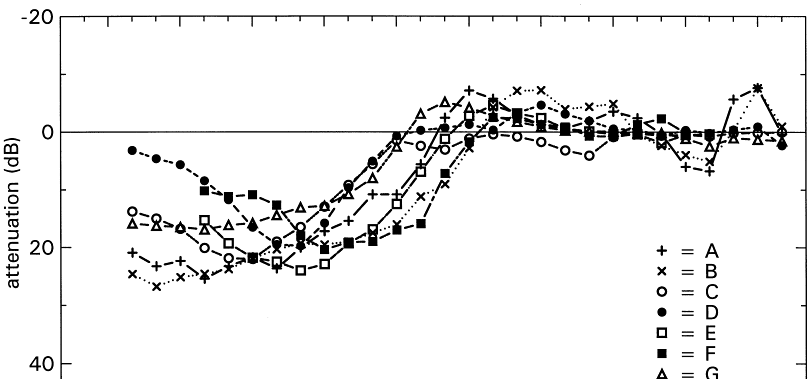

Fig. 12. Comparison of the active attenuation of commercial ANR systems.

The curves clearly indicate that most systems provide an additional attenuation of 10–15 dB in a frequency range between 80 and 800 Hz. Only systems A, B and E offer a much higher attenuation. For systems A-B an additional 6 dB stability range is included. This is unknown for the other systems. But the negative attenuation values indicate the same stability.

6. Conclusions

Feed-back based ANR-system are normally designed for a stable operation. This leads however to reduced performance. Therefore, it is attractive to design an adaptive algorithm witch automatically optimises the system parameters for different application.

Digital ANR allows for dynamic optimisation of system performance in order to eliminate the effect of user dependent acoustical conditions under the earmuff or the type of ambient noise.

Various algorithms were developed which automatically tune some system parameters as loop gain and compensation network response. The criteria used for this optimisation are: sound attenuation, level at the ear, speech intelligibility, and an adaptation of the speech signal for users with a slight hearing loss. Monitoring of the noise dose is also performed by a specific algorithm.

References

[1] LUEG P., *Process of silencing sound oscillations*, US Patent No. 2043416 (1936).

[2] NELSON P.A., ELLIOTT S.J., *Active control of sound*, Academic Press, London 1993.

[3] OLSON H.F., MAY E.G., *Electronic sound absorber*, J. Acoust. Soc. Am., **25**, 1130–1136 (1953).

[4] STEENEKEN H.J.M., HOUTGAST T., *A physical method for measuring speech-transmission quality*, J. Acoust. Soc. Am., **67**, 318–326 (1980).

[5] STEENEKEN H.J.M., VERHAVE J., *Personal active noise reduction with integrated speech communication devices: development and assessment*, Proc. Agard lecture series Copenhagen 1996.

[6] STEENEKEN H.J.M., *Personal active noise reduction with integrated speech communication devices: development and assessment*, Noise and Health, **1**, 67–75 (1998).

[7] ZERA J., BRAMMER A.J., PAN G.J., *Comparison between subjective and objective measures of active hearing protector and communication headset attenuation*, J. Acoust. Soc. Am., **101**, 6, 3486–3497 (1997).