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MEASUREMENTS ON AGC HEARING INSTRUMENTS BY USE OF BROAD-BAND TEST SIGNALS AND A PSYCHOACOUSTICAL MODEL

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ABSTRACT

Estimation of frequency curves and Speech Transmission Indices have been performed on six hearing instruments using broad-band test signals. All the instruments had some sort of signal dependent signal processing implemented. The test signals consisted of recordings of modulated and unmodulated noise signals and a female and a male voice. The estimation was based on a simpel model of auditory perception using a filter bank of Gamma-Tone filters to simulate the frequency selectivity of the ear. Test signals processed by a hearing instrument and unprocessed test signals are compared at model outputs to get the desired frequency curve estimates. Speech Transmission Indices was estimated from correlations between corresponding filter output signals. It is shown that this correlation technique (as well as the conventional coherence technique) has obvious disadvantages when the measured system uses signal dependent signal processing.

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INTRODUCTION

Measurements on hearing instruments are done according to the standard IEC-118. The standard describes how to measure frequency curves, distortions and other characteristics of an instrument. Measurements according to this standard have successfully been used since many years for technical assessments on hearing instruments.

A limitation in the standard is, however, that it is not possible to measure frequency curves of hearing instruments which change their characteristics depending on the input signal, e.g. AGC-instruments (AGC, Automatic Gain Control). To an increasing extent hearing instrument manufacturers are incorporating noise reducing circuits in their instruments. Such instruments make decisions, the outcome of which depends on input signal statistics, on which signal processing scheme to use.

A possible way to measure such instruments is to use speech or speech-like signals as test signals. Measures of frequency curves and distortions could then be defined based on cross-correlation techniques and the coherence function.

Such measurements on hearing instruments have been performed by Dyrlund (1989, 1991) and others.

Within the American National Standards Institute, Committee S3, Bioacoustics, there is work going on to establish a new standard for measurements on hearing instruments. The methods are based on broad-band noise test signals and cross-correlation technique (ANSI, 1991).

Measuring systems using the cross-correlation technique are mainly aiming at a purely technical description of the system under test. Another objective of measurements is to estimate the function of a hearing instrument from a perceptual point of view. A measuring system which uses a psychoacoustical model to define a distortion measure has been developed at the Laboratory of Acoustics in Lyngby, Denmark (Ludvigsen et al, 1990) and tested on a number of nonlinear hearing instruments. Speech Transmission Index theory (Humes et al, 1986; Pavlovic, 1987) was used to predict the intelligibility of the processed test signals. Good agreement between measured characteristics and results of psycho-acoustical listening tests were shown.

In this report a method to estimate frequency curves, distorsions and STI-values by use of a simple model of auditory perception is described. Test signals processed by a hearing instrument and unprocessed test signals are compared at model outputs to get the desired estimates.

The work is a part of greater Nordic cooperation work, where identical test signals and hearing instruments have been used, but where the methods of estimation were different.

MATERIAL AND METHODS

Hearing instruments

Six different hearing instruments (Table I) were used in this investigation. Three (H1, H3, H4) were ordinary AGC-instruments. One of these (H4) has a fixed output-controlled AGC which only works at high levels. Two (H2, H5) were signal processing instruments with automatic bass cut. One hearing instrument (H6) had a "Zeta Noise Blocker" circuit (Graupe et al, 1986 a,b).

Table I. Hearing instruments			
H1	Danavox 125-1	Input-controlled AGC	
H2	Bosch 33 PP-ANR	Automatic low-frequency noise reduction	
Н3	Oticon E35F	Output-controlled AGC	
Н4	Widex ES6	Output-controlled AGC	
H5	Bosch 33PP-ANC	Automatic low-frequency noise control	
H6	Maico SP137	Zeta Noise Blocker	

Test signals

The five test signals shown in Table II were taken from a DAT-tape produced by Project Odin at the Acoustical Laboratory, Technical University of Denmark. All signals on the tape had the same long-term RMS value.

Table II.	Test	signals	
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- P2 Speech-shaped noise
- P4 Modulated speech-shaped noise
- P6 Speech-shaped noise with modulation from P10
- P8 Male voice
- P10 Female voice

These test signals were sampled at 20 kHz, lowpass filtered at 7 kHz and stored on computer.

Equipment and measurements

The actual measurements on the hearing instruments took place in an anechoic room. The frequency response of the loudspeaker system is shown in Fig 1. In order to equalize the frequency response of the loudspeaker system, the test signals were digitally filtered with an inverse filter. The total, equalized loudspeaker response is shown in Fig 2.

The recording and playback system was implemented on an IBM/AT compatible computer and a front-end-processor (TAMP3), developed at the Department of Technical Audiology. TAMP3 has a TMS32010 signal-processor, AD/DA converters, tracking anti-aliasing filters, a controllable attenuator and a preamplifier.

The test signals were output to the loudspeaker system from the DA-converter and received by the hearing instruments. The output signals from an occluded ear simulator (B&K 4157) attached to the hearing instruments were A/D-converted and stored on computer hard disc. This was done for the five test signals and for the six hearing instruments at full-on-gain and at a reduction of gain of 10, 20 and 30 dB. During the recordings the AGC controls were set to their most active positions. Reference signals were also recorded. The presentation level was 70 dB SPL.

In order to calculate the frequency responses,

distorsions and STI-values the recorded signals from the hearing instruments were compared with the reference signals after being processed through channels A and B respectively according to the signal processing scheme in Fig. 3. The scheme was implemented on a DSP card (Loughborough Sound Images DSP32C PC System Board). Each channel in the scheme constitutes a simple model of the auditory periphery.

Auditory model

Each of the two filter banks in the model consists of 29 bandpass filters. The filters were implemented as complex "Gamma-Tone"-filters of second order (Patterson, 1988). The frequency characteristics of such filters are in good agreement with the shapes of PTC-curves (Psychoacoustical Tuning Curve) of normal-hearing subjects.

Fig. 4 shows the frequency response for the filter centered at 1 kHz and Fig. 5 shows the frequency responses of all filters. According to Moore & Glasberg (1983) the equivalent rectangular bandwidth (ERB) of an auditory filter in the ear is

ERB(f) =
$$6.23*10^{-6}*f_0^2 + 93.39*10^{-3}*f_0 + 28.52$$

(1000<10000)

where f_0 is the frequency in Hz.

The complex time discret Gamma-Tone filter of order two has a complex double-pole. The transfer function is thus given by

$$G(z) = \begin{bmatrix} 1 - r_0 \\ 1 - z^{-1}r_0e^{j\theta_0} \end{bmatrix}^2$$

where $r_0e^{j\theta}0$ is the coordinate for the poles. The corresponding complex pulse response is given by

$$g(n) = (n+1)r_0^n e^{jn\theta} 0$$

The ERB of this filter is given by

ERB_r =
$$\pi \frac{1 - r_0}{1 + r_0} \left[1 + \left[\frac{1 - r_0}{1 + r_0} \right]^2 \right]$$
 (radians)

or

$$ERB_{\Gamma} = \frac{f_{s}}{2} * \frac{1 - r_{0}}{1 + r_{0}} \left[1 + \left[\frac{1 - r_{0}}{1 + r_{0}} \right]^{2} \right]$$
(Hz)

where f_{S} is the sampling frequency. As r_0 is close to 1 the second term inside the brackets can be omitted with only minor influence on the result. The last formula can then be rewritten

$$r_0 = \frac{f_s - 2ERB_{\Gamma}}{f_s - 2ERB_{\Gamma}}$$

If ERB for the auditory filter is substituted for ERB_Γ we get a relation between r_0 and center frequency of the filter. The pole angle θ_0 can be found from

$$\theta_0 = 2\pi f_0/f_s$$

The center frequencies of adjacent filters are separated by one half of the equivalent rectangular bandwidth.

For each complex filter the intensity was calculated as the square of the complex output signal filtered through a lowpass filter of first order with a time constant of 125 ms.

Frequency curves were estimated as the square root of the quotient between the intensity of a filter in channel A (with the input signal processed by a hearing instrument) and the intensity of the corresponding filter in channel B (with the reference signal as input signal) as a function of the filter center frequency.

The intensities of corresponding filters in the two filterbanks were correlated to obtain a measure of correlated vs uncorrelated energi (S/N-ratio) as a function of filter center frequency (see Appendix A).

STI-calculations

The STI-calculations were done according to Fig. 3. To the S/N value in dB in each frequency band 12 dB was added. The sum was then divided by 30 and the quotient was limited to the range zero to one. Each

band has an individual importance factor according to Pavlovic (1987), see Table III, which the quotient was multiplied by. The results for all bands were added to give the STI-value.

Table III. Importance factor Filter frequency Importance in Hz 110.671 0.003 151.533 0.008 0.013 196.483 0.019 245.955 300.438 0.025 360.482 0.032 0.040 426.703 499.796 0.043 580.550 0.043 669.856 0.043 768.731 0.043 0.043 878.336 0.043 1000.000 1135.257 0.043 1285.879 0.043 0.043 1453.928 0.043 1641.815 1852.374 0.043 2088.964 0.043 2355.590 0.043 2657.068 0.043 2999.243 0.043 0.043 3389.273 3836.027 0.043 4350.625 0.040 0.034 4947.200 5644,001 0.028 6465.008 0.021 7442.377 0.013

RESULTS

The estimated frequency gain curves for the different hearing instruments and test signals are shown in figs 6 - 29. Curves are shown for full-ongain and gain reductions of 10, 20 and 30 dB. Instruments H1, H2, H5 and H6 have gain curves with magnitudes controlled by the volume control. Instruments H3 and H4 which have output-controlled AGC, give gain curves which are less controlled by volume control settings. This is especially true for H3 (figs 14-17), for which the gain has to be reduced by 30 dB before the magnitude of the gain curves starts to decrease. Instrument H4 has a fixed AGC which is active only at the highest levels. For all the instruments tested there is a dependence of test signal on the gain curves. For instrument H4 (Figs. 18-21) this is true only for the highest gain. This difference in gain curves occurs because the signal adaptive circuits in the instrument are active. It is most obvious for instruments H1 and H3.

The correlations as a function of filter frequency between instrument processed and unprocessed test signals are found in figs 30-53. In general these correlations are low. (A correlation of 0.7 means a S/N-ratio of 0 dB). There are exceptions, however: Instrument H4 (except at highest gain), and instrument H6. Even these two instruments have low correlations for some of the test signals, although.

The equivalent S/N-ratio as a function of filter frequency were calculated according to Appendix A. The S/N-ratio together with the importance factor of the filterband, determined the STI-value. STI-

values for the various hearing instruments as a function of test signals are shown in figs 54-59. The gain reduction is shown as a parameter in the figures. Instruments H2 and H5 seemes to be least influenced by different gain values. This is also true for instrument H1 with the exception of test signal P2. Instrument H3 is most influenced by gain settings. Instrument H4 gives rather high STIvalues except for test signal P2, and for all programs at 0 dB gain reduction. The highest STIvalues are obtained for instruments H4 (except for 0 dB gain reduction) and H6 for test signal P6. The overall impression is that different instruments behave quite differently for the various test signals. This is clearly demonstrated in Figs. 60-64 which show STI-values as a function of hearing instrument for the various test signals. In Fig. 65 the mean of the STI-values as a function of test signal are shown, calculated over hearing instruments and gain reductions. There is, on the average, no great difference between the STI-values obtained from dissimilar test signals. Fig. 66, finally, gives the overall STI-values of the instruments averaged over gains and test signals. Instrument H6 has got a somewhat higher STIvalue than the other instruments.

DISCUSSION

In linear system theory the coherence function is widely used to judge the validity of a measured frequency response since disturbances like noise and distortion products tend to decrease the

coherence. If it is known that the noise level is low, it is natural to regard the decrease in coherence as caused by nonlinearities and from this decrease define a measure of total distortion.

Anyway, for the purpose of calculating the Articulation Index or the Speech Transmission Index, it is straightforward to treat this decrease in coherence simply as caused by an uncorrelated noise source and define a S/N-ratio as the quotient of coherent power to uncoherent power. The arguments for using correlations between model filter outputs as a basis for calculations of S/N-ratios and STI-values are the same as the arguments for using the coherence function.

Unfortunately, as is shown in Appendix B, the correlations will be reduced for a system with AGC or similar types of signal processing if for instance the gain is changed over the time of measurement. This is the case for the instruments measured in this experiment, possibly with the exception of instrument H6 and H4 with gain reductions of 10, 20 and 30 dB.

What is shown in Appendix B is also true for coherence measurements since the coherence function is the correlation between two signals evaluated as a function of frequency.

The obvious conclusion of the Appendix B is that correlation or coherence techniques do not separate the nonlinear distortion caused by memoryless nonlinearities from distortion caused by gain variations. If we want to distinguish between distortions from memoryless nonlinearities which cause perceptually more harm and distortions from

purposely introduced AGC-type nonlinearities we have to find other means to achieve this goal.

It can also be shown that, when using a modelbased approach as done in this work, a slopeing of the frequency curve of a hearing instrument within a filter band will further decrease the correlation as may be seen in most of the correlation figures for higher frequencies.

ACKNOWLEDGEMENTS

The author wants to express his gratitude to Göran Lundberg for his help with some of the programming and simulation tasks involved in the project. I also want to thank the other participants in the Nordic working group, namely Ole Dyrlund, Carl Ludvigsen (who also selected and prepared the test signals) and Torben Poulsen.

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APPENDIX A

Given two signals x(t) and y(t), which are output signals from corresponding filters in channels A and B resp of Fig. 3, we want to estimate the amount of energy in signal y(t) which is dependent on x(t). For simplicity we assume that

$$E\{x(t)\} = E\{y(t)\} = 0$$
 (1)

where E{'} is espected value. We first form estimates of the intensities of the signals by

$$I_{x}(t) = LP\{x^{2}(t)\}$$
 (2)

and

$$I_{y}(t) = LP\{y^{2}(t)\}$$
 (3)

respectively, where LP{'} is a lowpass operator. It can be observed that

$$E\{I_x(t)\} = \sigma_x^2$$
 and $E\{I_y(t)\} = \sigma_y^2$ (4)

where ${\sigma_x}^2$ and ${\sigma_y}^2$ are the variance of signal x(t) and y(t) respectively. Our signal model is

$$I_{\mathbf{Y}}(t) = a * I_{\mathbf{X}}(t) + I_{\mathbf{V}}(t)$$
 (5)

where $I_{V}(t)$ is a noise component. If we take the expected values of (5) we get with the use of (4)

$$\sigma_{y}^{2} = a*\sigma_{x}^{2} + \sigma_{v}^{2} \tag{6}$$

The signal-to-noise ratio (S/N) is given by

$$S/N = \frac{a*\sigma_x^2}{\sigma_v^2} = \frac{a*\sigma_x^2}{\sigma_v^2 - a*\sigma_x^2}$$
 (7)

If we cross-correlate $I_{\mathbf{X}}(t)$ and $I_{\mathbf{Y}}(t)$ we get

$$r = E\{(I_{x}(t) - \sigma_{x}^{2})(I_{y}(t) - \sigma_{y}^{2})\} =$$

$$= E\{I_{x}(t)I_{y}(t) - \sigma_{x}^{2}\sigma_{y}^{2}\}$$
(8)
$$r = E\{I_{x}(t)(a*I_{x}(t)+I_{y}(t))\} - \sigma_{x}^{2}(a*\sigma_{x}^{2}+\sigma_{y}^{2}) =$$

$$= a*(E\{I_{x}^{2}(t)\} - \sigma_{x}^{4})$$
(9)

From (9) we get

$$a = \frac{r}{E\{I_x^2(t)\} - \sigma_x^4}$$
 (10)

With (10) and (8) equation (7) can be written

$$S/N = \frac{\sigma_{x}^{2} * E\{I_{x}(t)I_{y}(t) - \sigma_{x}^{2}\sigma_{y}^{2}\}}{\sigma_{y}^{2} * [E\{I_{x}^{2}(t)\} - \sigma_{x}^{4}] - \sigma_{x}^{2} * E\{I_{x}(t)I_{y}(t) - \sigma_{x}^{2}\sigma_{y}^{2}\}}$$
(11)

APPENDIX B

Given two channels A and B with signals $y_A(t)$ and $y_B(t)$ which are of the form

$$y_A(t) = a(t)*x(t)$$

and
 $y_B(t) = x(t)$ respectively,

where a(t) is a slowly varying function depending upon x(t), we want to calculate the correlation between the channels. The signals could for instance be the output and input signals to an AGC circuit. The squared correlation is estimated by

$$r^{2} = \frac{\left[\frac{1}{T}\int_{T_{0}}^{T_{0}+T} a(t)|x(t)|^{2}dt\right]^{2}}{\frac{1}{T}\int_{T_{0}}^{T_{0}+T} |a(t)x(t)|^{2}dt\frac{1}{T}\int_{T_{0}}^{T_{0}+T} |x(t)|^{2}dt}$$

Schwarz's inequality yields

$$\left[\int_{a}^{b} f(x) g(x) dx \right]^{2} \leq \int_{a}^{b} [f(x)]^{2} dx \int_{a}^{b} [g(x)]^{2} dx$$

Identifying f(x) with a(t)x(t) and g(x) with x(t) gives the result

$$r^2 \leq 1$$

with equality only if a(t) is a constant, i.e. perfect correlation can only be achieved if the gain is constant.

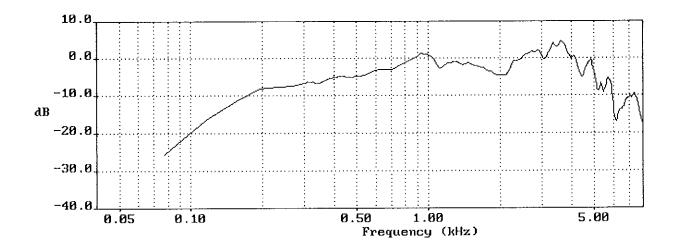


Fig. 1. Frequency response of loudspeaker system.

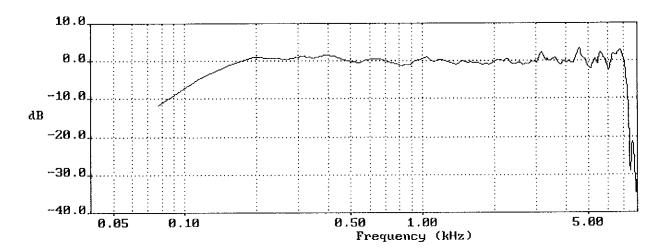


Fig. 2. Equalized frequency response of loudspeaker system.

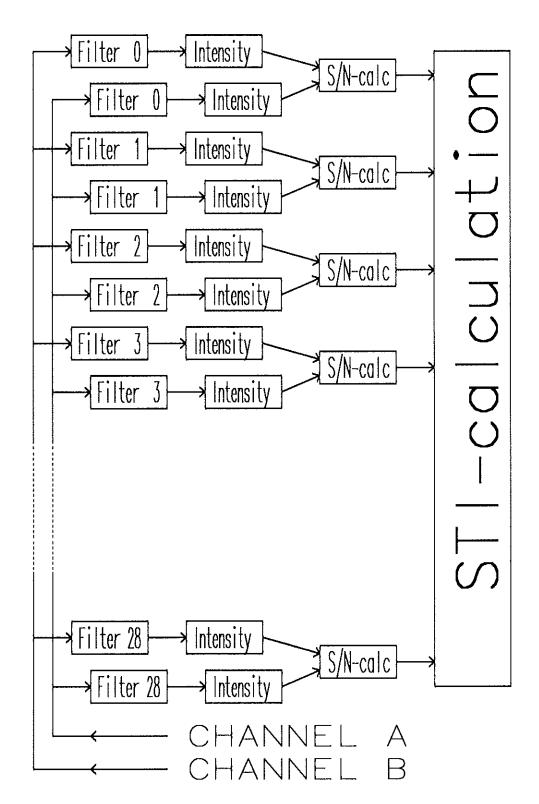


Fig. 3. Signal processing scheme for calculation of STI-value.

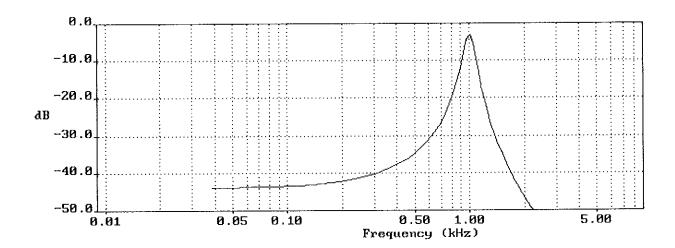


Fig. 4. Frequency response of Gamma-Tone filter with center frequency 1 kHz.

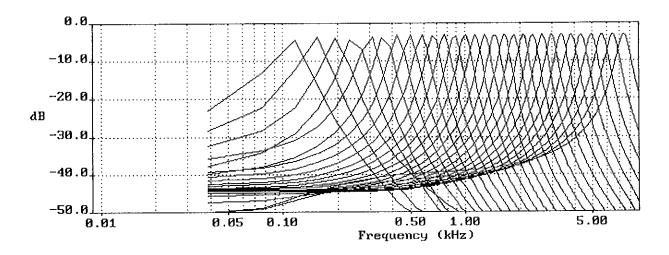


Fig. 5. Frequency responses of Gamma-Tone filters in each filter bank.

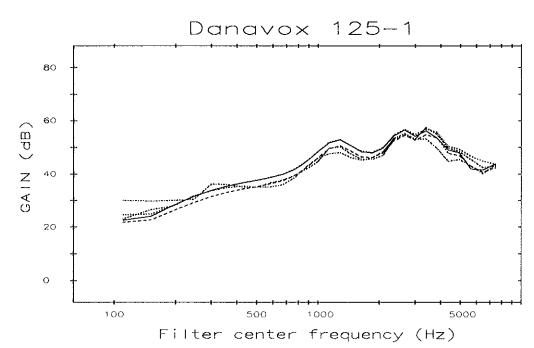


Fig. 6. Gain as a function of filter center frequency for hearing instrument H1 for the different programs. Gain reduction 0 dB.

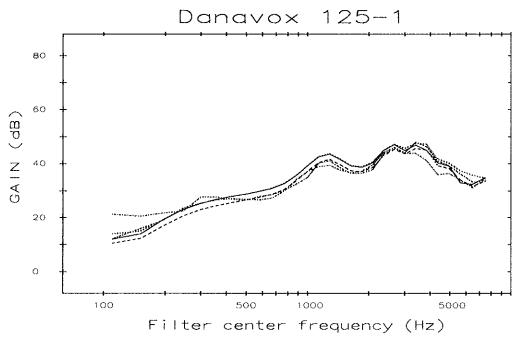


Fig. 7. Gain as a function of filter center frequency for hearing instrument H1 for the different programs. Gain reduction 10 dB.

	P2
	P4
~~~~~~	₽6
	<b>P8</b>
	P 10

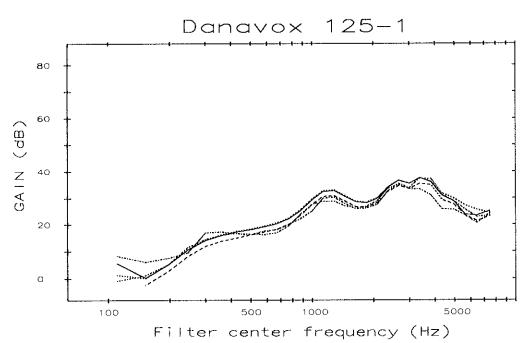


Fig. 8. Gain as a function of filter center frequency for hearing instrument H1 for the different programs. Gain reduction 20 dB.

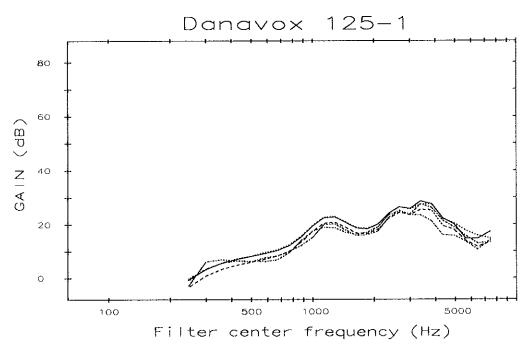


Fig. 9. Gain as a function of filter center frequency for hearing instrument H1 for the different programs. Gain reduction 30 dB.

Annal Contra	₽2
	P4
	P6
	Р8
	P10

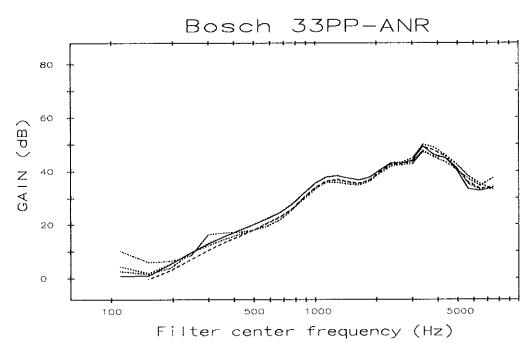


Fig. 10. Gain as a function of filter center frequency for hearing instrument H2 for the different programs. Gain reduction 0 dB.

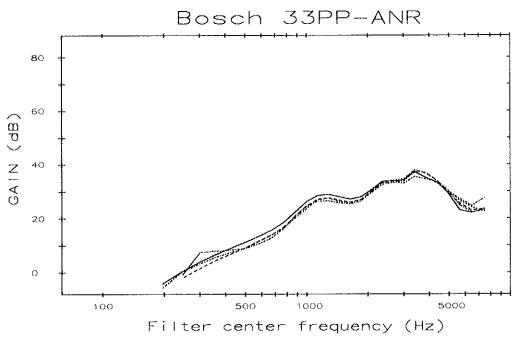


Fig. 11. Gain as a function of filter center frequency for hearing instrument H2 for the different programs. Gain reduction 10 dB.

P2
 P4
 P6
 P8
 P10

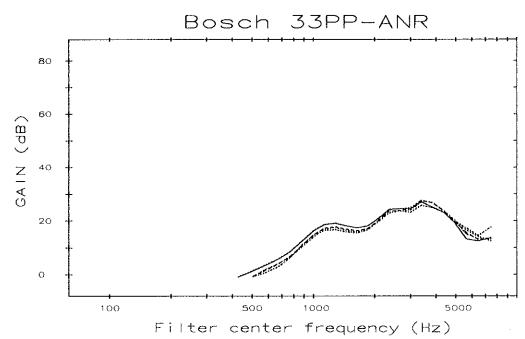


Fig. 12. Gain as a function of filter center frequency for hearing instrument H2 for the different programs. Gain reduction 20 dB.

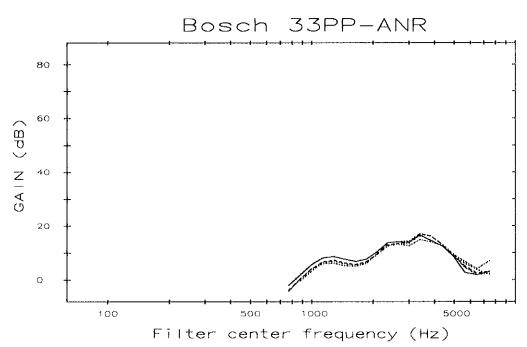


Fig. 13. Gain as a function of filter center frequency for hearing instrument H2 for the different programs. Gain reduction 30 dB.

	P2
,	₽4
	P6
	89
	P10

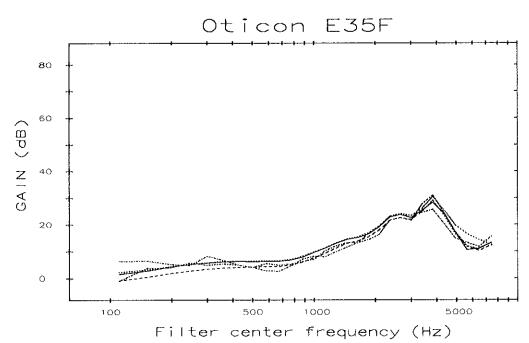


Fig. 14. Gain as a function of filter center frequency for hearing instrument H3 for the different programs. Gain reduction 0 dB.

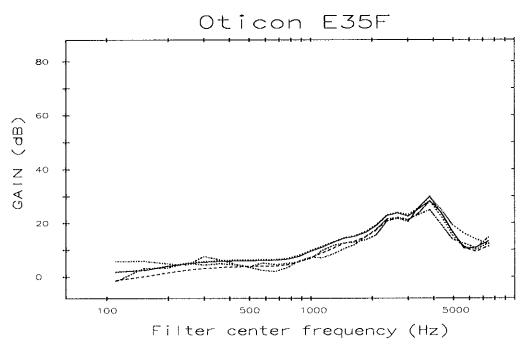


Fig. 15. Gain as a function of filter center frequency for hearing instrument H3 for the different programs. Gain reduction 10 dB.



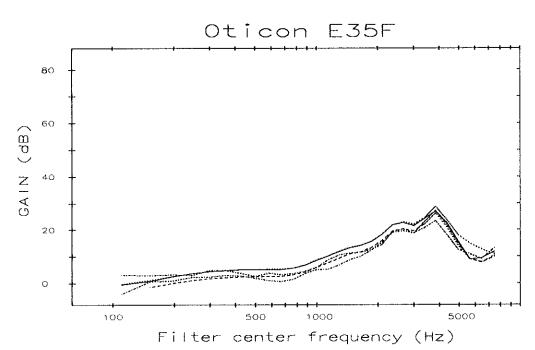


Fig. 16. Gain as a function of filter center frequency for hearing instrument H3 for the different programs. Gain reduction 20 dB.

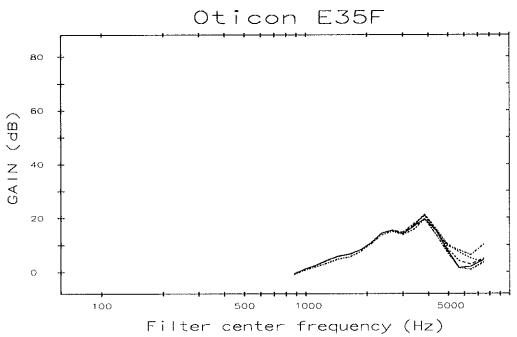


Fig. 17. Gain as a function of filter center frequency for hearing instrument H3 for the different programs. Gain reduction 30 dB.

	P2
,	P4
	<b>P</b> 6
	89
	P10

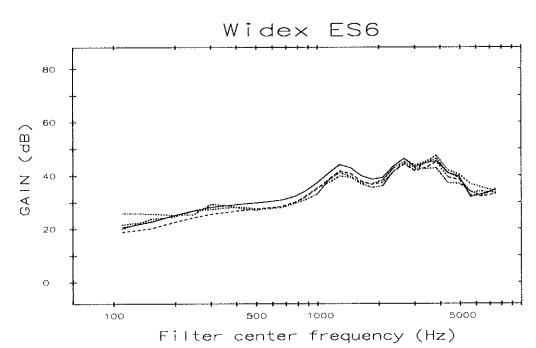


Fig. 18. Gain as a function of filter center frequency for hearing instrument H4 for the different programs. Gain reduction 0 dB.

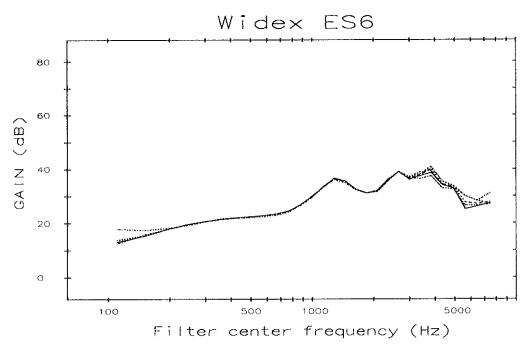


Fig. 19. Gain as a function of filter center frequency for hearing instrument H4 for the different programs. Gain reduction 10 dB.

	65
,	₽4
	Р6
	P8
	P10

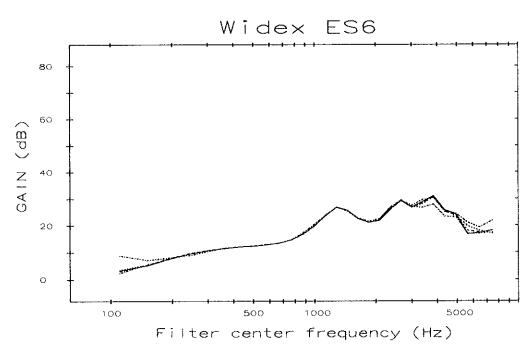


Fig. 20. Gain as a function of filter center frequency for hearing instrument H4 for the different programs. Gain reduction 20 dB.

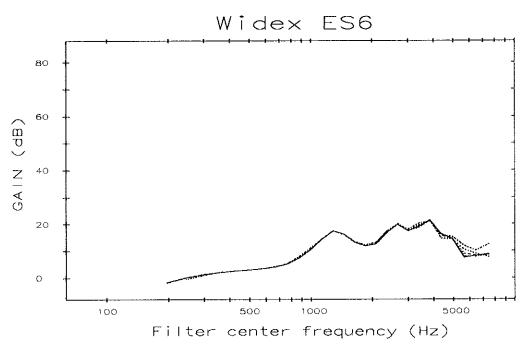


Fig. 21. Gain as a function of filter center frequency for hearing instrument H4 for the different programs. Gain reduction 30 dB.

 P2
 P4
 Р6
 P8
 P10

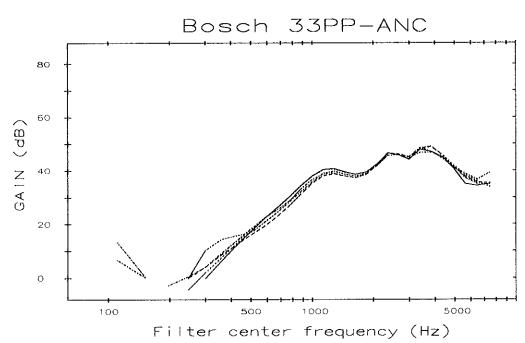


Fig. 22. Gain as a function of filter center frequency for hearing instrument H5 for the different programs. Gain reduction 0 dB.

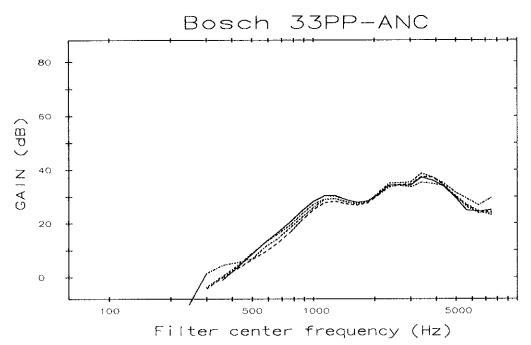


Fig. 23. Gain as a function of filter center frequency for hearing instrument H5 for the different programs. Gain reduction 10 dB.



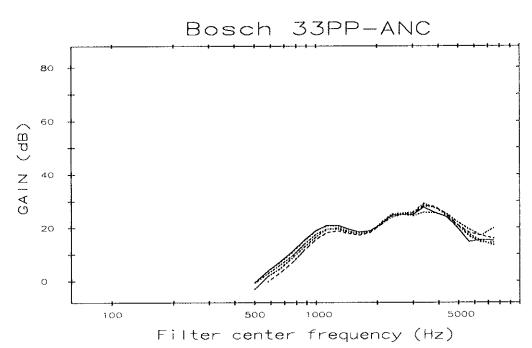


Fig. 24. Gain as a function of filter center frequency for hearing instrument H5 for the different programs. Gain reduction 20 dB.

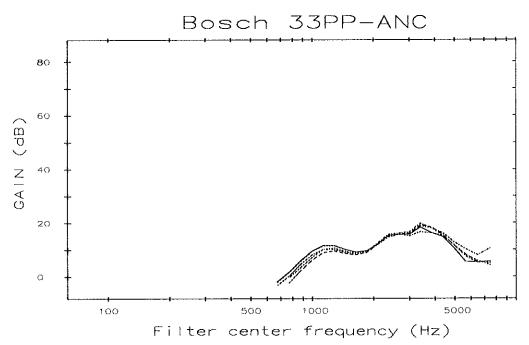


Fig. 25. Gain as a function of filter center frequency for hearing instrument H5 for the different programs. Gain reduction 30 dB.



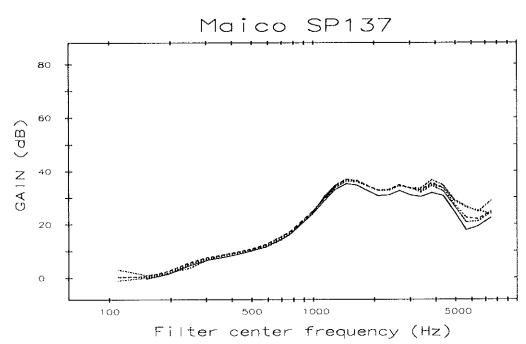


Fig. 26. Gain as a function of filter center frequency for hearing instrument H6 for the different programs. Gain reduction 0 dB.

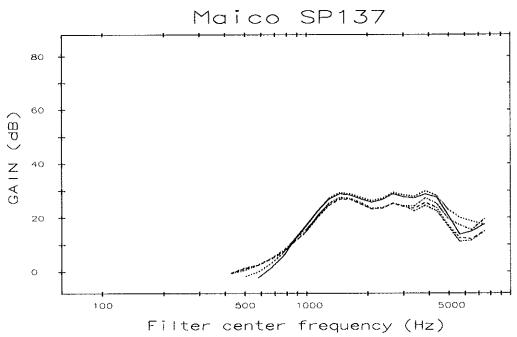


Fig. 27. Gain as a function of filter center frequency for hearing instrument H6 for the different programs. Gain reduction 10 dB.



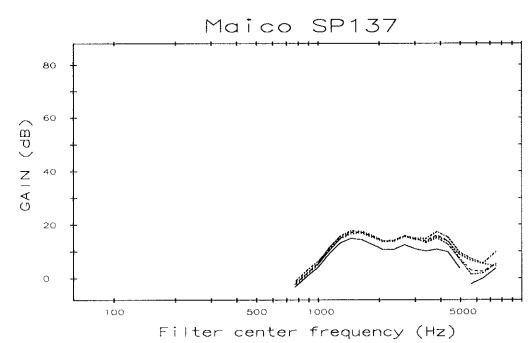


Fig. 28. Gain as a function of filter center frequency for hearing instrument H6 for the different programs. Gain reduction 20 dB.

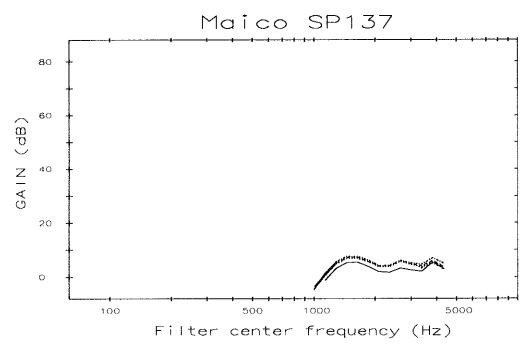


Fig. 29. Gain as a function of filter center frequency for hearing instrument H6 for the different programs. Gain reduction 30 dB.



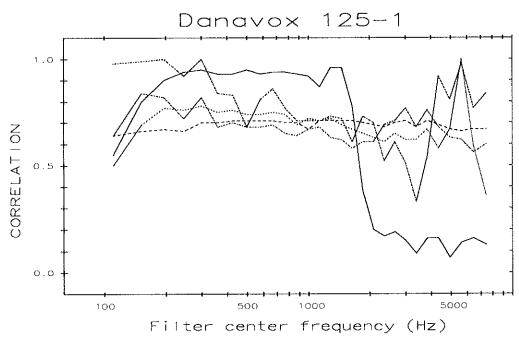


Fig. 30. Correlation from auditory model as a function of filter frequency for hearing instrument H1 for the different programs. Gain reduction 0 dB.

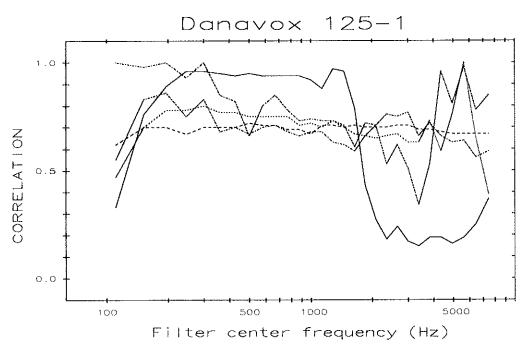


Fig. 31. Correlation from auditory model as a function of filter frequency for hearing instrument H1 for the different programs. Gain reduction 10 dB.

 ₽2
 ₽4
 Р6
 Р8
 P10

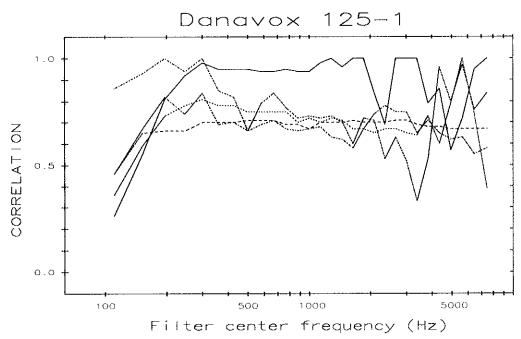


Fig. 32. Correlation from auditory model as a function of filter frequency for hearing instrument H1 for the different programs. Gain reduction 20 dB.

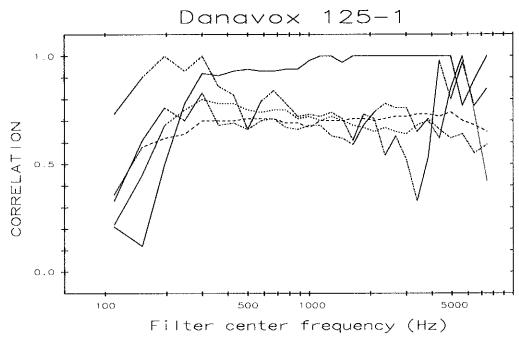


Fig. 33. Correlation from auditory model as a function of filter frequency for hearing instrument H1 for the different programs. Gain reduction 30 dB.

	P2
	ρ4
× * * * * * * * * * * * * * * * * * * *	Р6
	Р8
.,	P10

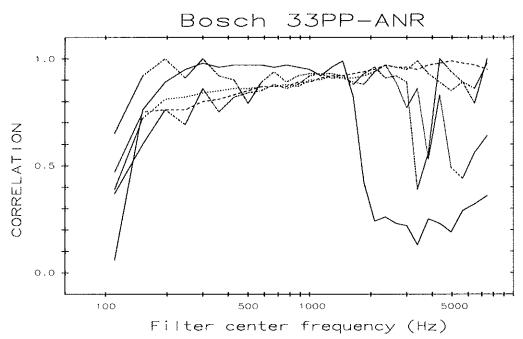


Fig. 34. Correlation from auditory model as a function of filter frequency for hearing instrument H2 for the different programs. Gain reduction 0 dB.

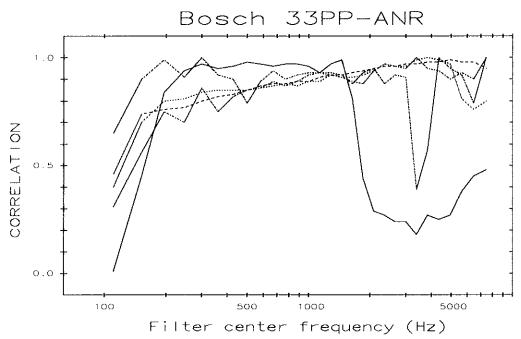


Fig. 35. Correlation from auditory model as a function of filter frequency for hearing instrument H2 for the different programs. Gain reduction 10 dB.

P2
 P4
 P6
 Р8
 P10

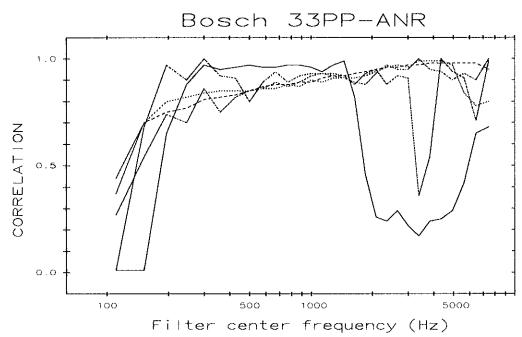


Fig. 36. Correlation from auditory model as a function of filter frequency for hearing instrument H2 for the different programs. Gain reduction 20 dB.

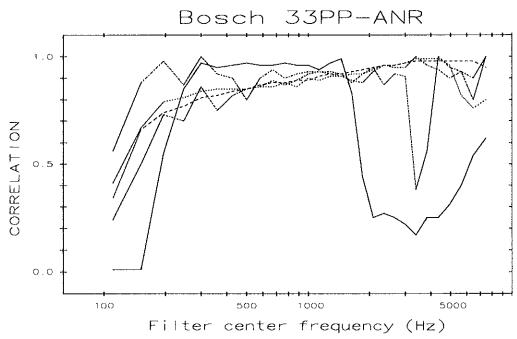


Fig. 37. Correlation from auditory model as a function of filter frequency for hearing instrument H2 for the different programs. Gain reduction 30 dB.



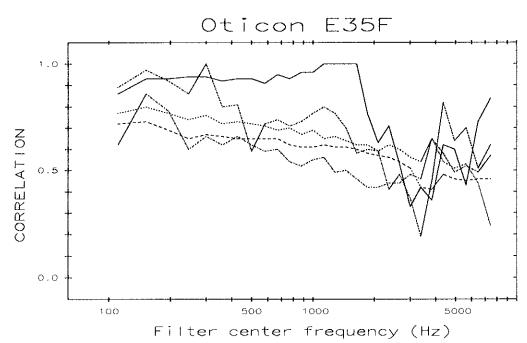


Fig. 38. Correlation from auditory model as a function of filter frequency for hearing instrument H3 for the different programs. Gain reduction 0 dB.

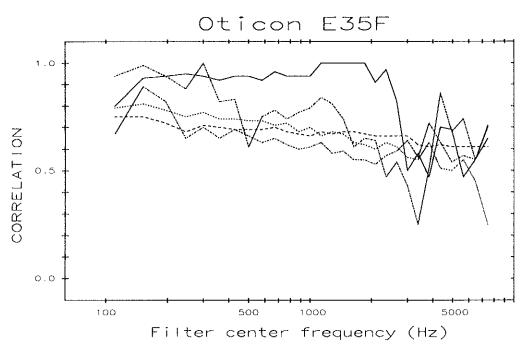


Fig. 39. Correlation from auditory model as a function of filter frequency for hearing instrument H3 for the different programs. Gain reduction 10 dB.

1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1	P2
	Ρ4
	P6
	P8
	₽10

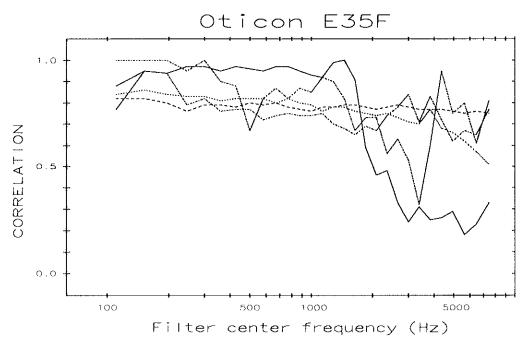


Fig. 40. Correlation from auditory model as a function of filter frequency for hearing instrument H3 for the different programs. Gain reduction 20 dB.

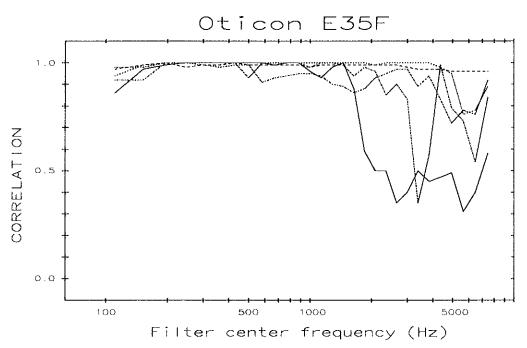


Fig. 41. Correlation from auditory model as a function of filter frequency for hearing instrument H3 for the different programs. Gain reduction 30 dB.

 P2
 P4
 P6
 PB
 ₽10

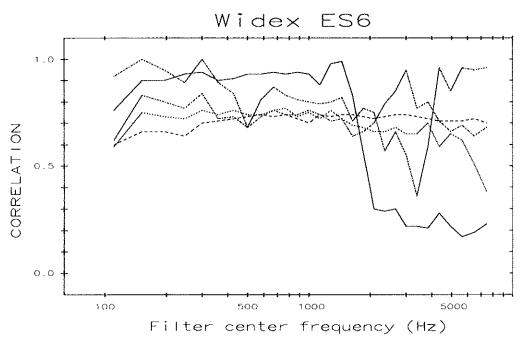


Fig. 42. Correlation from auditory model as a function of filter frequency for hearing instrument H4 for the different programs. Gain reduction 0 dB.

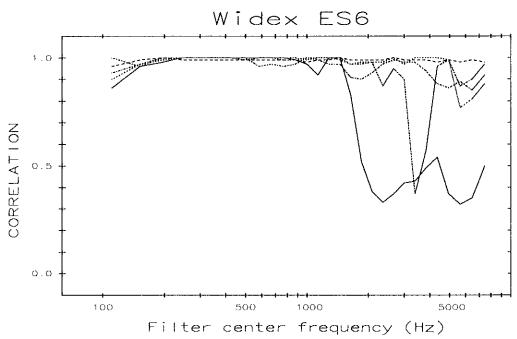


Fig. 43. Correlation from auditory model as a function of filter frequency for hearing instrument H4 for the different programs. Gain reduction 10 dB.

 P2
 ₽4
 156
 89
 P10

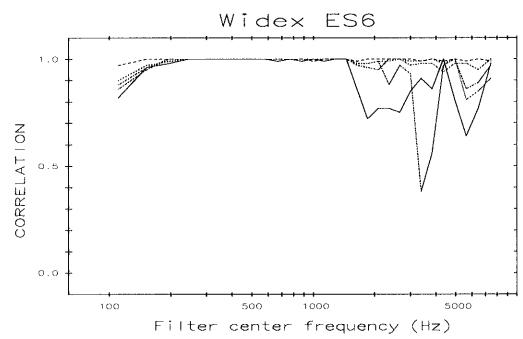


Fig. 44. Correlation from auditory model as a function of filter frequency for hearing instrument H4 for the different programs. Gain reduction 20 dB.

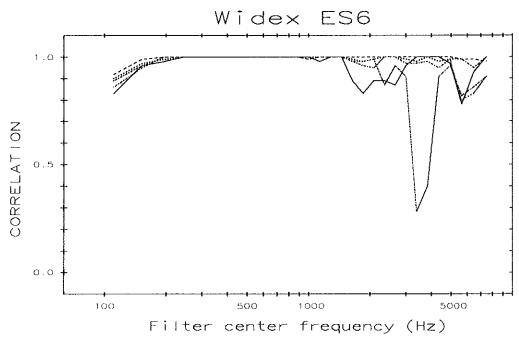


Fig. 45. Correlation from auditory model as a function of filter frequency for hearing instrument H4 for the different programs. Gain reduction 30 dB.

 ₽2
 ₽4
 P6
 P8
 P10

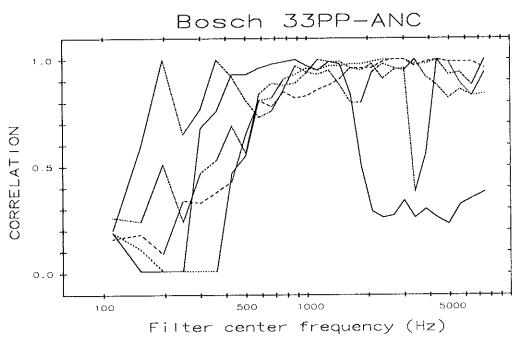


Fig. 46. Correlation from auditory model as a function of filter frequency for hearing instrument H5 for the different programs. Gain reduction 0 dB.

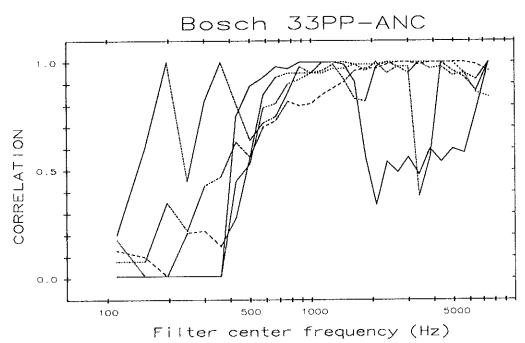


Fig. 47. Correlation from auditory model as a function of filter frequency for hearing instrument H5 for the different programs. Gain reduction 10 dB.

 P2
 P4
 P6
 Р8
 P10

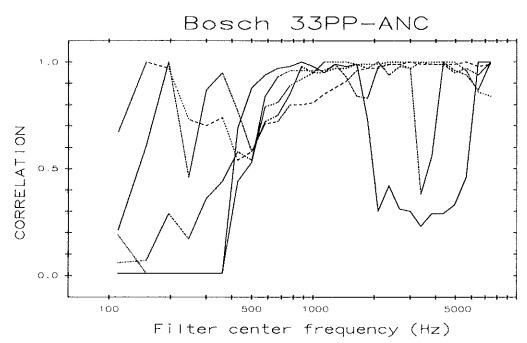


Fig. 48. Correlation from auditory model as a function of filter frequency for hearing instrument H5 for the different programs. Gain reduction 20 dB.

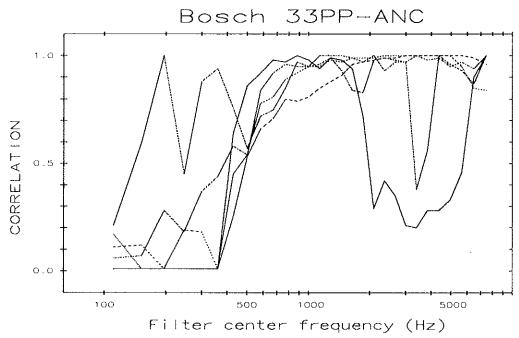


Fig. 49. Correlation from auditory model as a function of filter frequency for hearing instrument H5 for the different programs. Gain reduction 30 dB.

	P2
	P4
	P6
	P8
. * . *	P10

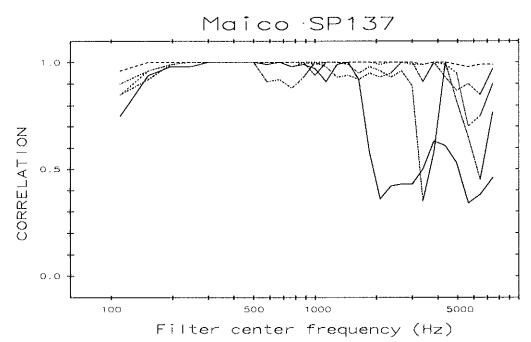


Fig. 50. Correlation from auditory model as a function of filter frequency for hearing instrument H6 for the different programs. Gain reduction 0 dB.

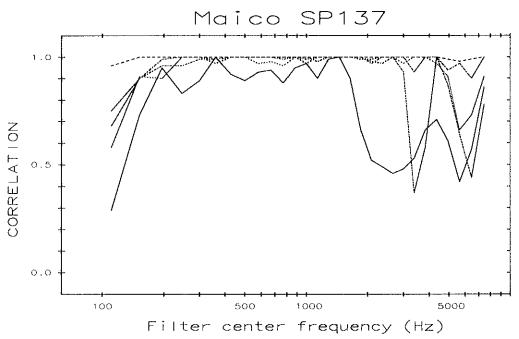


Fig. 51. Correlation from auditory model as a function of filter frequency for hearing instrument H6 for the different programs. Gain reduction 10 dB.



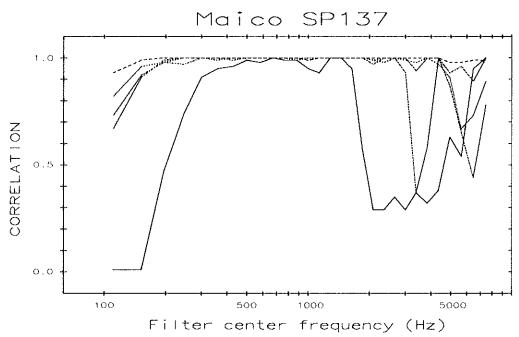


Fig. 52. Correlation from auditory model as a function of filter frequency for hearing instrument H6 for the different programs. Gain reduction 20 dB.

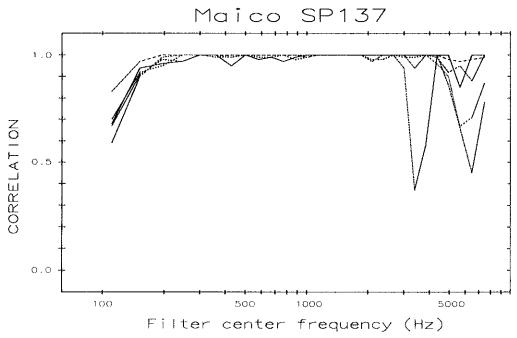


Fig. 53. Correlation from auditory model as a function of filter frequency for hearing instrument H6 for the different programs. Gain reduction 30 dB.



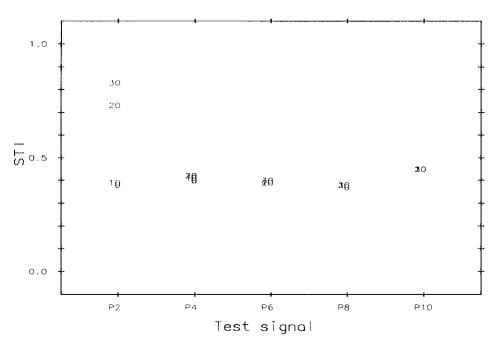


Fig. 54. STI as a function of test signal for hearing instrument H1. Gain reduction from max gain as parameter.

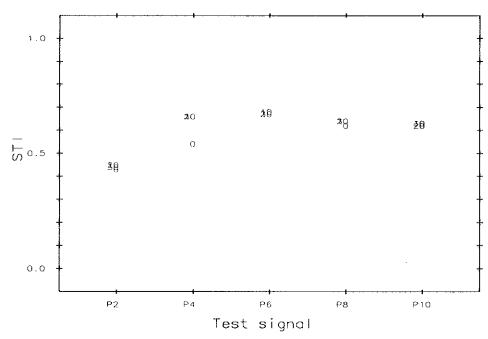


Fig. 55. STI as a function of test signal for hearing instrument H2. Gain reduction from max gain as parameter.

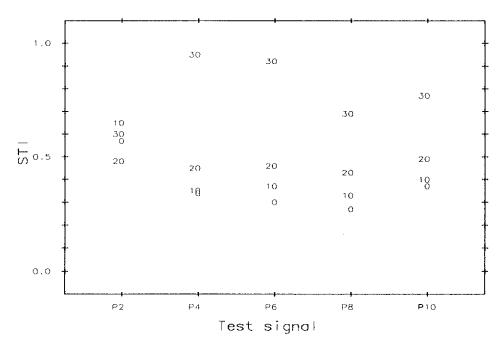


Fig. 56. STI as a function of test signal for hearing instrument H3. Gain reduction from max gain as parameter.

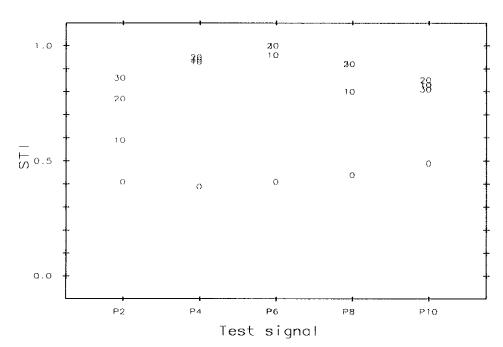


Fig. 57. STI as a function of test signal for hearing instrument H4. Gain reduction from max gain as parameter.

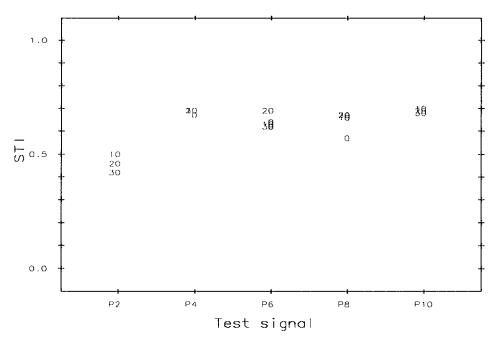


Fig. 58. STI as a function of test signal for hearing instrument H5. Gain reduction from max gain as parameter.

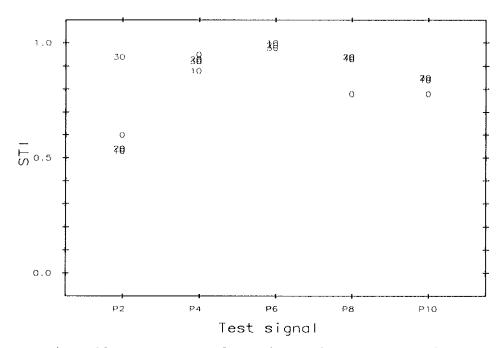


Fig. 59. STI as a function of test signal for hearing instrument H6. Gain reduction from max gain as parameter.

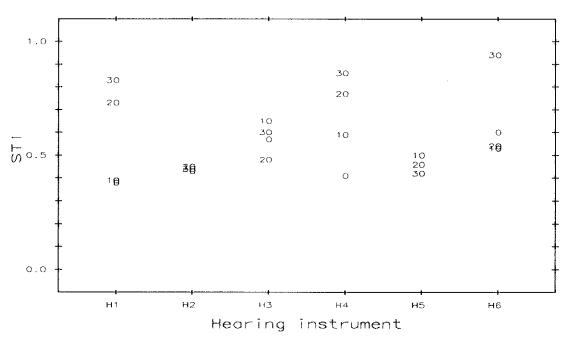


Fig. 60. STI as a function of hearing instrument for test signal P2. Gain reduction from max gain as parameter.

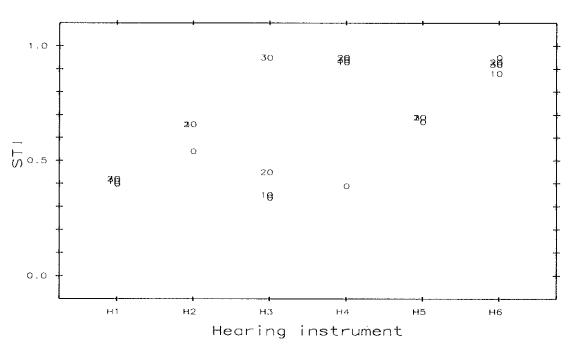


Fig. 61. STI as a function of hearing instrument for test signal P4. Gain reduction from max gain as parameter.

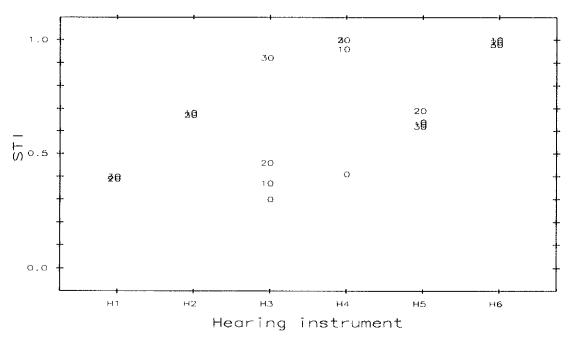


Fig. 62. STI as a function of hearing instrument for test signal P6. Gain reduction from max gain as parameter.

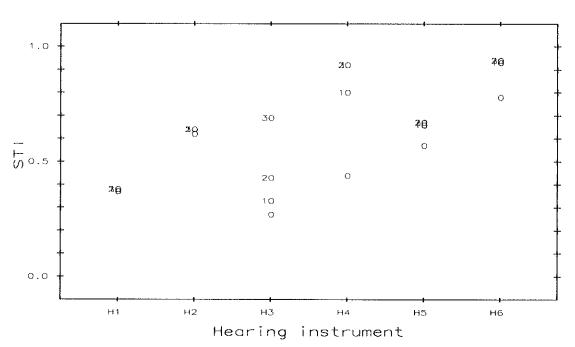


Fig. 63. STI as a function of hearing instrument for test signal P8. Gain reduction from max gain as parameter.

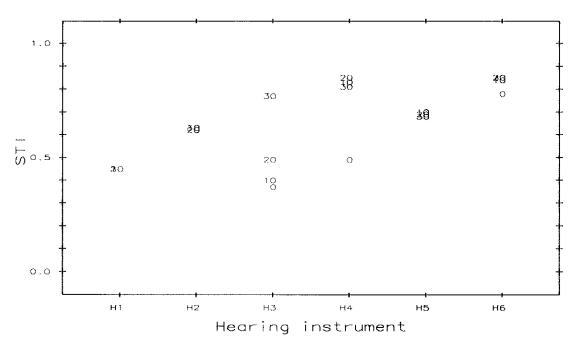


Fig. 64. STI as a function of hearing instrument for test signal P10. Gain reduction from max gain as parameter.

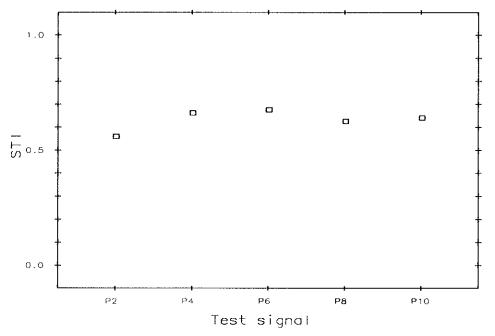


Fig. 65. STI as a function of test signal. Mean values over hearing instruments and gains.

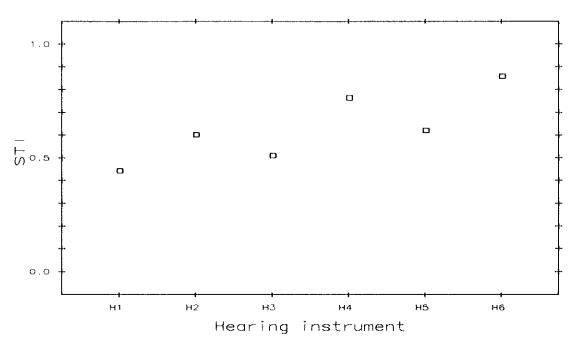


Fig. 66. STI as a function of hearing instrument. Mean values over test signals and gains.

TA101	1981	Sentences for testing speech intelligibility innoise. B Hagerman.
TA100	1981	Modulation transfer function and speech transmission index - a method to predict speech intelligibility in Swedish? - The Swedish contribution to the IEC 29B/WG15 multilanguage test.  B Hagerman, A-C Lindblad.
TA99	1980	En metod för subjektiv utvärdering av "kompaktanläggningar". B Lindström.
TA98	1980	Clinical assessment of perceived sound quality in hearing aids. A Gabrielsson, B Hagerman, C Berg, A Ovegård, L Anggård.
TA97	1980	Statistical analysis of fricative sounds. A Olofsson.
<b>TA</b> 96	1980	A method for suppression of periodic noise signals. S-A Frykholm, A Olofsson.
*TA95	1980	Hörapparaters frekvenskurvor: IEC-coupler, KEMAR, människa. U Olsson.
TA94	1980	Some ideas in computerized acoustical impedance measurements. S-A Frykholm.
TA93	1979	Assessment of perceived sound quality in high fidelity sound-reproducing systems. A Gabrielsson, S-A Frykholm, B Lindström.
*TA92	1978	Statistical treatment of data from listening tests on sound-reproducing systems. A Gabrielsson.
<b>TA91</b>	1978	Taluppfattbarhet för högtalarsystem vid Arlanda utrikesterminal. B Hagerman, A-C Lindblad.
TA90	1978	A generalization of the concept of total harmonic distortion. A Olofsson.
*TA89	1978	Ett programpaket för mätning, syntes och analys av linjära tidskontinuerliga och tidsdiskreta system. Å Olofsson.
88AT	1978	Reliability in the determination of speech reception threshold (SRT). B Hagerman.
та87	1978	Detection of nonlinear distortion in telephone systems. A Gabrielsson, A-C Lindblad, B Lindström, A Olofsson.
*TA86	1977	Adjective ratings and dimension analyses of perceived sound quality of headphones. A Gabrielsson, S-A Frykholm, H Sjögren.
*TA85	1977	Adjective ratings and dimension analyses of perceived sound quality of hearing aids. III. $\lambda$ Gabrielsson, H Sjögren.
*TA84	1977	Automatic gain control and hearing aids. The influence of different attack and release times on speech intelligibility for hearing impaired with recruitment. T Ahren, S Arlinger, C Holmgren, L Jerlvall, B. Johansson, A-C Lindblad, H Sjögren.
*TA83	1976	Detection of amplitude distortion by normal hearing and hearing impaired subjects. A Gabrielsson, P-O Nyberg, H Sjögren, L Svensson.
*TA82	1976	Preferred listening levels and perceived sound quality at different sound levels in "high fidelity" sound reproduction. A Gabrielsson, H Sjögren.
*TA81	1976	Preliminärrapport slingor. R Ingelstam.
TA80	1975	Emulator för kommunikation mellan minidator typ Honeywell 316 och stordator typ IBM 360. Bo Jacobsson.
TA79	1975	Mätning av amplitud- och faskurvor för hörapparater med hjälp av minidator. Å Olofsson.
TA78	1975	Styr- och registreringsenhet för psykoakustisk mätplats. Å Ölofsson, L Persson, H Sjögren.
*TA77	1975	Adjective ratings and dimension analyses of perceived sound quality of hearing aids. II. A Gabrielsson, H Sjögren.
*TA76	1975	Similarity ratings and dimension analyses of perceived sound quality of hearing aids. A Gabrielsson, H Sjögren.
*TA75	1974	Adjective ratings and dimension analyses of perceived sound quality of hearing aids. I. A dabrielsson, H Sjögren.
TA74	1974	Assessment of comfort and discomfort levels for pure tone, a methodological study. A Gabrielsson, B Johansson, Bodil Johansson, A-C Lindblad, L Persson.