# Auditory filter shapes at low center frequencies

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(Received 26 July 1989; accepted for publication 2 February 1990)

Auditory-filter shapes were estimated in normally hearing subjects for signal frequencies ($f_s$) of 100, 200, 400, and 800 Hz using the notched-noise method [R. D. Patterson and I. Nimmo-Smith, J. Acoust. Soc. Am. 67, 229–245 (1980)]. Two noise bands, each $0.4f_s$ wide, were used; they were placed both symmetrically and asymmetrically about the signal frequency to allow the measurement of filter shape and asymmetry. Two overall noise levels were used: 77 and 87 dB SPL. In deriving the shapes of the auditory filters, account was taken of the nonflat frequency response of the Sennheiser HD424 earphone, and also of the frequency-dependent attenuation produced by the middle ear. The auditory filters were asymmetric; the upper skirt was steeper than the lower skirt. The asymmetry tended to be greater at the higher noise level. The equivalent rectangular bandwidths (ERBs) of the filters at the lower noise level had average values of 36, 47, 87, and 147 Hz for values of $f_s$ of 100, 200, 400, and 800 Hz, respectively. The standard deviations of the ERBs across subjects were typically about 10% of the ERB values. The signal-to-masker ratio at the output of the auditory filter required to achieve threshold increased markedly with decreasing $f_s$.

PACS numbers: 43.66.Dc, 43.66.Cb [NFV]

## INTRODUCTION

The traditional critical bandwidth (CB) function flattens off below 500 Hz at a value of about 100 Hz (Zwicker, 1961; Scharf, 1970; Zwicker and Terhardt, 1980). However, when this function was proposed, there were few direct estimates of the CB at low center frequencies, and the form of the function at low frequencies was strongly influenced by measurements of the critical ratio. The critical ratio is now known not to provide reliable estimates of the critical bandwidth (Patterson and Moore, 1986; Moore, 1987). Furthermore, some of the early data indicated that the CB continued to decrease as the center frequency decreased below 500 Hz (Greenwood, 1961).

Moore and Glasberg (1983) published a summary of measurements of the bandwidth of the auditory filter obtained using notched-noise and rippled-noise maskers (see Patterson and Moore, 1986, for a review). Although there is no universally accepted definition of the CB, most researchers view it as a measure of the effective bandwidth of the auditory filter. The equivalent rectangular bandwidths (ERBs) of the auditory filters described by Moore and Glasberg continued to decrease as the center frequency decreased below 500 Hz, reaching a value of about 40 Hz for a center frequency of 125 Hz.

Since 1983, there have been several studies of the CB at low center frequencies (Schorer, 1986; Fastl and Schorer, 1986; Rosen and Stock, 1989). All of these studies show that the CB continues to decrease at frequencies below 500 Hz, but the extent of the decrease differs across studies. Some of the discrepancies can be attributed to differences in the way in which the data are analyzed (see the comment by Moore, following the paper of Fastl and Schorer, 1986). Fastl and Schorer suggested that differences in the frequency responses of the earphones used were also a source of discrepancies. Specifically, earphones used with free-field equalizers appeared to give higher CB estimates at low frequencies than earphones without free-field equalizers. However, this cannot account for the small CB values obtained by Fidell *et al.* (1983) at low center frequencies, since their study was conducted using loudspeakers in an anechoic chamber. A third possible source of discrepancies is the strong variation of absolute threshold with frequency at low center frequencies. This could have different effects on different psychoacoustical measures, and the effects might vary with the overall levels used.

In the present experiments we estimated the shapes of the auditory filters at low center frequencies (100–800 Hz) using the notched-noise technique (Patterson and Moore, 1986). Conditions were included where the notch was positioned asymmetrically about the signal frequency, so as to allow the measurement of auditory-filter asymmetry (Patterson and Nimmo-Smith, 1980). There have been few previous measurements of auditory-filter asymmetry at low center frequencies. Fidell *et al.* (1983) reported measurements using asymmetric notched noise for two subjects at a

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center frequency of 124 Hz, but they did not derive auditory filter shapes from the data. In analyzing our results, the frequency response of the earphone used was taken into account, and particular attention was paid to the influence of absolute threshold.

## I. METHOD

### A. Stimuli

Each masker consisted of two bands of noise, one above and one below the signal frequency ($f_s$). The value of $f_s$ was 100, 200, 400, or 800 Hz. Each band had a width of $0.4f_s$, except when this would have caused components to fall below 0 Hz; the bandwidth was then reduced accordingly. The deviation of the nearer edge of each noise band from $f_s$ is expressed relative to $f_s$ and denoted by $\Delta$. For $f_s = 200$, 400, and 800 Hz, the pairs of values of $\Delta$ for the lower and upper bands were: 0.0 and 0.0, 0.1 and 0.1, 0.2 and 0.2, 0.3 and 0.3, 0.4 and 0.4, 0.5 and 0.5, 0.6 and 0.6, 0.3 and 0.5, 0.4 and 0.6, 0.5 and 0.7, 0.6 and 0.8, 0.5 and 0.3, 0.6 and 0.4, 0.7 and 0.5, and 0.8 and 0.6, respectively. In pilot experiments it was established that bands whose upper edges were at 40 Hz or below did very little masking. Therefore, at $f_s = 100$ Hz, the pairs of values of $\Delta$ for the lower and upper bands were: 0.0 and 0.0, 0.1 and 0.1, 0.2 and 0.2, 0.3 and 0.3, 0.4 and 0.4, 0.5 and 0.5, 0.1 and 0.3, 0.2 and 0.4, 0.3 and 0.5, 0.4 and 0.6, 0.5 and 0.7, 0.3 and 0.1, 0.4 and 0.2, 0.5 and 0.3, and 0.6 and 0.4, respectively.

The noise bands were created by passing an analog noise source through a digital filter (IHR Universal Filter). They were recorded on a hi-fi VHS video recorder (Mitsubishi model HS-413UR) for playback during the experiment (this was done because the IHR filter was located in a different laboratory from the one used for testing). The noise was gated using a Wilsonics electronic gate (model BSIT), set to give 10-ms cosine ramps. On each trial three bursts of noise were used, with 400-ms steady-state portions and 300-ms interstimulus intervals. Two nominal overall noise levels were used. The lower one, 77 dB SPL, was chosen to give masked thresholds well above absolute threshold for all notch widths, while keeping the loudness of the noise at a moderate value. The higher level, 87 dB, was used mainly to provide a comparison with results for hearing-impaired subjects (Peters and Moore, 1990). Such subjects usually have to be tested at relatively high noise levels, to ensure that masked thresholds are above absolute threshold. The lower noise level gave spectrum levels of 58, 55, 52, and 49 dB (re: 20 $\mu$Pa) for $f_s = 100$, 200, 400, and 800 Hz, respectively. In cases where the noise bandwidth was reduced (to avoid components falling below zero frequency), the spectrum level was held constant, rather than the overall level.

The signal was generated digitally using a 12-bit digital-to-analog converter (DAC) at a 5-kHz sampling rate. The output of the DAC was low-pass filtered using two Krohn Hite filters (model 3550), set to give a $-3$-dB point of 2000 Hz. The signal had 10-ms cosine ramps, and a steady-state portion of 200 ms. The signal occurred at random in one of the three masker bursts. It was gated on 200 ms after the start of the masker, and terminated synchronously with the masker. The levels of the signal and masker were controlled by Wilsonics programmable attenuators. After attenuation, the signal and masker were combined in an active adder, and passed through a final manual attenuator to one earpiece of a Sennheiser HD424 headset, chosen for its smooth low-frequency response.

Absolute thresholds were measured for each subject using 200-ms signals at frequencies of 50, 75, 100, 150, 200, 300, 400, 500, and 800 Hz.

### B. Procedure

An adaptive three-interval three-alternative forced-choice method was used. A three-down one-up rule was used to estimate the signal level corresponding to the 79.4% correct point on the psychometric function (Levitt, 1971). The signal level was initially changed in 8-dB steps. After two turnpoints the step size was decreased to 4 dB, and after two further turnpoints to 2 dB. Twelve turnpoints were obtained and threshold was taken as the mean of the levels at the last eight turnpoints. For some subjects, only eight turnpoints were obtained, and the levels at the last four were used to estimate threshold. When the standard deviation of the eight (or four) levels exceeded 5 dB, that run was discarded. At least two threshold estimates were obtained for each condition. If the two estimates differed by more than 2 dB, at least one additional estimate was obtained, and all estimates were averaged. Observation intervals were marked by lights on the response box, and feedback was provided by a light indicating the correct interval. Absolute thresholds were estimated using the same method. Subjects were tested individually in a single-walled sound-attenuating chamber.

### C. Subjects

Ten subjects were tested in total. All subjects had absolute thresholds less than 20 dB HL at all audiometric frequencies, and no history of hearing disorders. Eight of the subjects were aged 20–25. The others had ages of 27 and 43. Subjects were given only a brief practice period before testing began. However, we have found that performance on this task changes little with practice. Five subjects were tested only at the lower noise level, four only at the higher noise level, and one at both noise levels.

### D. Calibration of the earphone

The earphone was calibrated in two ways. In the first, we used a Knowles XL7093 miniature microphone, positioned at the entrance to the ear canal. The calibration curve supplied with the microphone showed its frequency response to be flat ($+/-0.25$ dB) between 20 and 5000 Hz. Measurements taken with five subjects showed a high degree of inter- and intrasubject reliability (standard deviation across subjects typically less than 1 dB) for frequencies up to 2000 Hz, the range of interest here. The response of the earphone was essentially flat ($\pm 1.5$ dB) from 100 to 2000 Hz. Relative to the response at 1000 Hz, the response was, on average, 3.8 dB down at 75 Hz, 8.6 dB down at 50 Hz, and 14.6 dB down at 30 Hz.

The second method of calibration was via the absolute thresholds of the subjects. The absolute thresholds were averaged and the form of the results compared with the normal absolute threshold of hearing published in ISO recommendation R.226 for free-field listening conditions (denoted by MAF, minimum audible field). For frequencies between 100 and 800 Hz, the average absolute threshold curve was parallel to the MAF curve, but shifted upwards by about 3 dB (this is to be expected, since the MAF curve applies to binaural listening with long-duration tones, whereas our subjects listened monaurally to 200-ms tones). Below 100 Hz, the curves diverged. The divergence indicated that the response of the earphones was 4.0 dB down at 75 Hz and 7.2 dB down at 50 Hz. This is in rather good correspondence with the measurements using the miniature microphone. In the rest of this paper, the calibration provided by the microphone is used, since this was obtained for a large number of closely spaced frequencies.

## II. CALCULATION OF AUDITORY-FILTER SHAPES

The method of deriving filter shapes from the data is similar to that described by Patterson and Nimmo-Smith (1980) and by Glasberg and Moore (1986), but with some significant modifications. These modifications are described in detail in Glasberg and Moore (1990), and we will give only brief descriptions here.

Each side of the auditory filter was assumed to have the form of the roex$(p,r)$ filter described by Patterson *et al.* (1982):

$$W(g) = (1-r)(1+pg)\exp(-pg) + r, \qquad (1)$$

where $g$ is the normalized deviation from the center of the filter (deviation from center frequency divided by center frequency), $p$ is a parameter determining the slope of the filter skirts, and $r$ a parameter that flattens the filter at frequencies remote from the center frequency, thereby placing a dynamic range limitation on the filter. The value of $p$ was allowed to differ for the upper and lower halves of the auditory filter; the upper and lower $p$ values are called $p_u$ and $p_l$, respectively. The value of $r$ was assumed to be the same for the two sides of the filter. The goodness of fit was not significantly improved by allowing $r$ to differ for the two sides of the filter.

The first problem in analyzing the results is to take into account the fact that the frequency response of the earphone used falls below 100 Hz. According to the power-spectrum model of masking, the threshold $P_s$ for a signal in noise is

$$P_s = K\int_0^\infty N(f)W(f)df, \qquad (2)$$

where $K$ is a constant, $f$ is frequency, $N(f)$ represents the long-term spectrum of the masker at the input to the filter, and $W(f)$ represents the intensity weighting applied by the auditory filter (Patterson and Moore, 1986); $K$ represents the signal-to-noise ratio required at the output of the filter to achieve threshold. When the noise spectrum is flat within its passbands, the signal threshold can be predicted by integrating the equation for the filter shape over the frequency range covered by the noise. This has formed the basis of earlier methods for estimating the auditory-filter shape, all of which made use of an analytic integral (e.g., Glasberg and Moore, 1986). However, if a frequency-dependent attenuation is applied before auditory filtering, the spectrum of the noise at the input to the auditory filter will not be flat within its passbands. To predict the signal threshold in this case, it is necessary to calculate the spectra of the stimuli at the input to the filter, and then to evaluate the integral in Eq. (2) by numerical methods. This is the method that we used to allow for the falling frequency response of the earphone.

A second problem is how to take into account the variation of absolute threshold with frequency, which is particularly marked at low center frequencies. Clearly, this can have a significant influence on measures of frequency selectivity. It is not entirely clear how to deal with this. One possibility is that, at least in young normally hearing listeners, the transducers within the cochlea are equally sensitive at all audible frequencies and the variation of absolute threshold with frequency reflects a frequency-dependent attenuation resulting from the transfer function of the outer and middle ear (see Dallos, 1973, and Pickles, 1988, for reviews). The auditory filter is usually conceived of as resulting from processes occurring after the middle ear. Hence, the variation of absolute threshold with frequency could reflect a frequency-dependent attenuation applied to all stimuli before auditory filtering takes place.

According to Dallos (1973), part of the sensitivity loss at low frequencies can be attributed to changes in the cochlear input impedance resulting from the action of the helicotrema (see, also, Nedzelnitsky, 1980). However, this can still be regarded as a frequency-dependent attenuation applied before auditory filtering. If this is the case, then the frequency-dependent attenuation should be taken into account in the fitting procedure for deriving filter shapes. Essentially, it can be handled in the same way as the correction for the frequency response of the earphone. In this paper, the attenuation applied is based on the shape of the MAF curve published in ISO recommendation R. 226. As described earlier, the mean absolute thresholds of our subjects as a function of frequency had the same form as this curve.

It seems likely, however, that part of the variation of absolute threshold with frequency arises in other ways. For frequencies below 1 kHz, the behavioral absolute threshold appears to change more rapidly with frequency than would be predicted from the transfer function of the outer and middle ear, both in cat (Lynch *et al.*, 1982) and in humans (Zwislocki, 1975). A possible explanation is that internal noise in the cochlea is greater at low frequencies than at high (Soderquist and Lindsey, 1972; Nedzelnitsky, 1980). This would raise absolute thresholds at low frequencies but would have little effect on the perception of low frequencies at high sound levels. This could explain why equal-loudness contours at high sound levels are flatter than the absolute threshold curve. The equal-loudness contours would be almost unaffected by the internal noise, and so their shape would largely reflect the transfer characteristic of the outer and middle ear. Hence, as well as using a frequency-dependent attenuation based on the MAF curve, we also used an

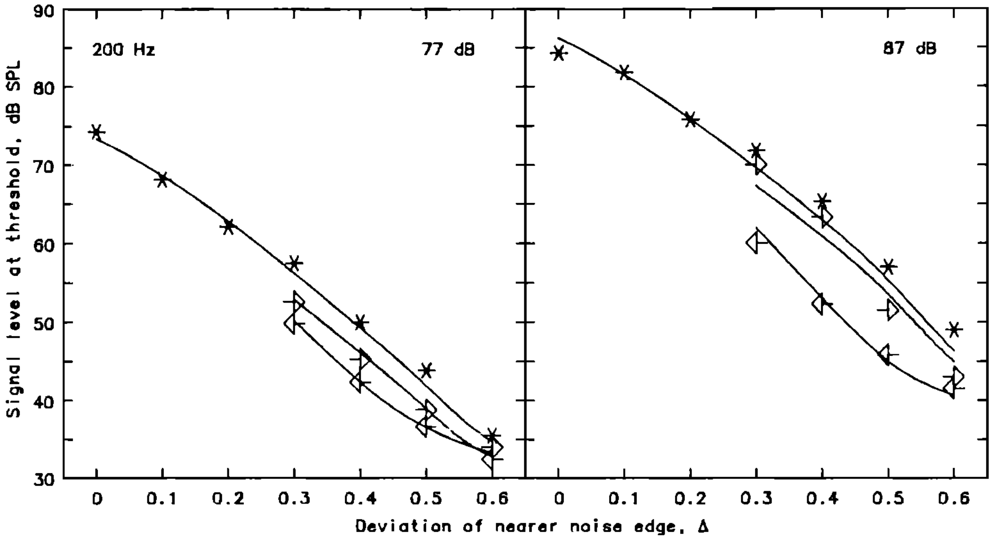


FIG. 1. Examples of individual data for a center frequency of 200 Hz, at the lower noise level (left panel) and the higher noise level (right panel). Thresholds are plotted as a function of the deviation of the nearer noise edge from $f_s$ (see text). Asterisks indicate conditions where the notch was symmetrically placed around $f_s$. Right-pointing arrows indicate conditions where the edge of the upper noise band was 0.2 units farther from $f_s$ than the edge of the lower band. Left-pointing arrows indicate conditions where the edge of the lower band was 0.2 units farther away. The lines indicate the thresholds fitted by the roex$(p,r)$ model using the ELC + Senn correction.

attenuation based on the 100-phon equal-loudness contour as specified in ISO recommendation R. 226.

In summary, the fitting procedure was modifed to allow filter shapes to be derived for cases where the noise spectra were subject to a frequency-dependent attenuation before reaching the auditory filter. In all cases, a correction for the frequency response of the Sennheiser earphone was included. Two different additional frequency-dependent attenuations were applied. One was based on the MAF curve; this case will be referred to as "MAF + Senn." The other was based on the 100-phon equal-loudness contour; this will be referred to as "ELC + Senn." For comparison purposes, we also derived auditory-filter shapes with the Sennheiser correction only ("Senn").

Two additional modifications to the fitting procedure should be mentioned. The procedure allows for the possibility of off-frequency listening; for each notch width the center frequency of the filter is allowed to shift so as to find the center frequency giving the highest signal-to-masker ratio. This shifted filter is used to predict the threshold for that notch width. In previous work (both our own and, as far as we are aware, that of others), the filter shape and bandwidth were assumed not to vary as a result of the shift. This is a reasonable assumption when the shift is small. However, in our data, the shift could sometimes be as large as $0.2f_s$ and, in a very few cases, was even greater. The first modification was that the shift was not allowed to be greater than $0.2f_s$. This had little effect in the majority of cases, but it prevented the anomalous fits that can sometimes occur when very large shifts are allowed (Glasberg and Moore, 1990). The second modification was to assume that the filter gets slightly sharper when it shifts down in center frequency, and slightly broader when it shifts up in center frequency. As described in Glasberg and Moore (1990), it was assumed that the equivalent rectangular bandwidth (ERB) of the auditory filter varies with center frequency according to the equation

$$\mathrm{ERB} = 24.7(4.37F + 1)\,, \tag{3}$$

where $F$ is the center frequency in kHz, and the ERB is in Hz. It was also assumed that the values of $p_u$ and $p_l$ vary with center frequency as 1/ERB.

## III. RESULTS AND DISCUSSION

### A. The effect of allowing for variations in sensitivity with frequency

We start by considering the effect on the derived filters of applying the various "corrections" described earlier. To do this we have selected two individual sets of data for a center frequency of 200 Hz, one at the lower noise level and one at the higher level. The derived filters for these data fall in the middle of the ranges for individual subjects (in terms of sharpness and asymmetry). Thus the data may be considered as typical of the whole data set. Figure 1 shows the two data sets and also the thresholds fitted by the roex$(p,r)$ model using the ELC + Senn correction. Table I gives the parameters of the filters derived from the data with no correction applied, and with the three different corrections applied: Senn; ELC + Senn; and MAF + Senn.

With no correction applied, the filter for the lower noise level is almost symmetric, while that for the higher level has a slightly shallower lower branch than upper branch. With the Senn correction applied the lower branches become shal-

TABLE I. Parameters of the auditory filters derived from two sets of individual data at 200 Hz, one at the lower noise level and one at the higher noise level. The filters were derived with no correction applied, and with three different corrections applied: Senn, ELC + Senn, and MAF + Senn. As well as giving the parameters defining the filter shapes ($p_l$, $p_u$, and $r$), the table gives the ERBs of the filters, and the value of $K$ (the signal-to-noise ratio at the output of the filter required to achieve threshold, in dB). The value of $r$ is expressed in dB. The ERB is given as a proportion of the center frequency.

| Noise level (dB) | Correction | $p_l$ | $p_u$ | $r$ | ERB | $K$ |
|---|---|---|---|---|---|---|
| 77 | none | 17.7 | 18.6 | −50 | 0.22 | −0.3 |
| | Senn | 17.1 | 18.3 | −46 | 0.23 | −1.0 |
| | ELC + Senn | 14.9 | 18.7 | −47 | 0.24 | −1.3 |
| | MAF + Senn | 11.3 | 20.0 | −50 | 0.28 | −1.9 |
| 87 | none | 16.9 | 21.7 | −59 | 0.21 | 3.2 |
| | Senn | 15.3 | 20.5 | −51 | 0.23 | 2.3 |
| | ELC + Senn | 13.3 | 21.1 | −52 | 0.25 | 1.7 |
| | MAF + Senn | 8.6 | 21.6 | −55 | 0.32 | 0.4 |

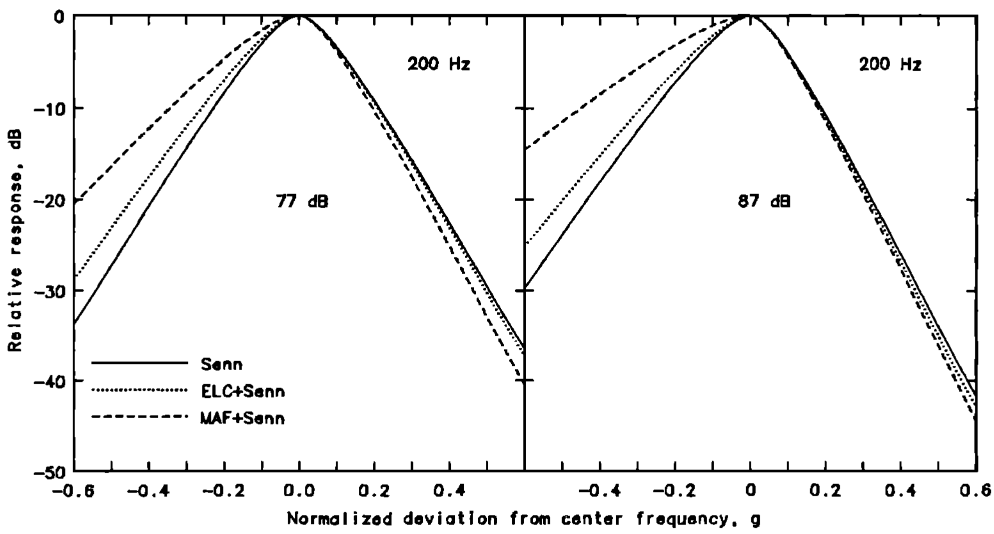


FIG. 2. Auditory-filter shapes derived from the data shown in Fig. 1, using three different corrections: Senn (solid lines), ELC + Senn (dotted lines), and MAF + Senn (dashed lines).

lower, but the change is small. Somewhat larger effects were found for a center frequency of 100 Hz. With the ELC + Senn correction, both filters become distinctly asymmetric, with shallower lower branches. With the MAF + Senn correction, the asymmetry is even more marked. The shapes of the derived filters for the Senn, ELC + Senn, and MAF + Senn corrections are shown in Fig. 2.

Over the entire data set, the goodness of fit of the roex($p$, $r$) model was similar for the ELC + Senn and MAF + Senn corrections; the root-mean-square (rms) deviation of the data from the fitted values was 1.8 dB. For the Senn correction the fit was slightly worse (rms deviation 1.9 dB), and, with no correction, it was worse still (rms deviation 2.1 dB). In general, the trends shown in Table I and Fig. 2 are representative of the whole data set. The major effect of applying the different corrections was on the low-frequency sides of the derived filters; the sharpness of the lower branches decreased with the ELC + Senn correction and decreased still further with the MAF + Senn correction. These effects were most marked for the two lowest center frequencies. In a few cases at $f_s = 100$ Hz, the MAF + Senn correction resulted in filters that were almost flat on the low-frequency side. This resulted in very large individual differences in the ERBs of the derived filters. The results using the ELC + Senn correction were much more uniform across subjects. It should be noted that, if the MAF + Senn correction were appropriate, then very little noise would have passed through the lower branch of the auditory filter at $f_s = 100$ Hz. Thus the filter shape would be poorly defined by the data.

These results are consistent with our earlier suggestion that the ELC + Senn correction is the most appropriate one to apply. This correction probably corresponds reasonably well to the frequency-dependent attenuation applied by the outer and middle ear. It gives more uniform results across subjects than the MAF + Senn correction, and it gives a better fit to the data than the Senn correction. In the rest of this paper, we will concentrate on the filters derived using the ELC + Senn correction.

## B. The shapes and bandwidths of the derived filters

Rather than plot all of the derived filter shapes, we selected three for each center frequency and level: the least sharp, the most sharp, and one close to the average. The data from which the "typical" filter shapes were derived are shown in Figs. 3–5, for center frequencies of 100, 400, and

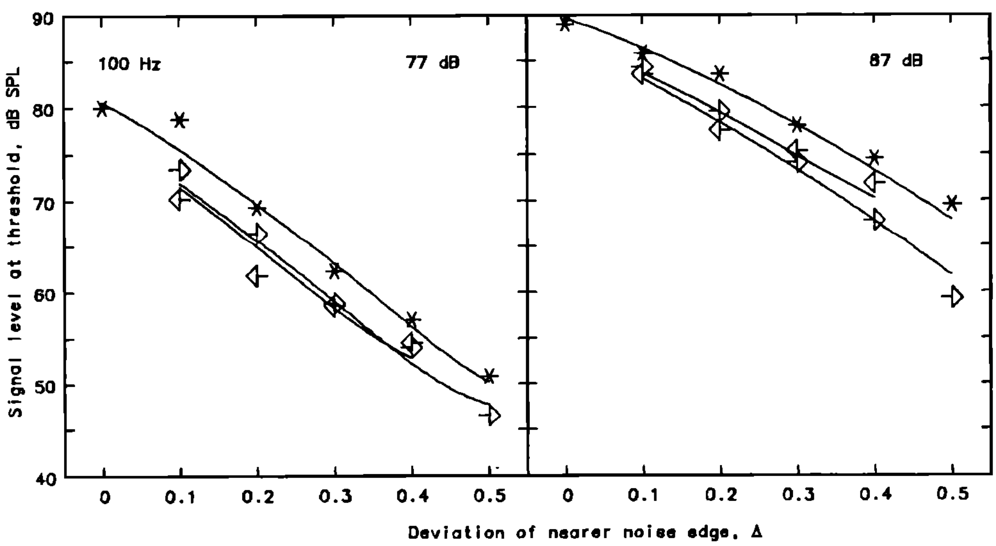


FIG. 3. As in Fig. 1, but showing typical results at $f_s = 100$ Hz.

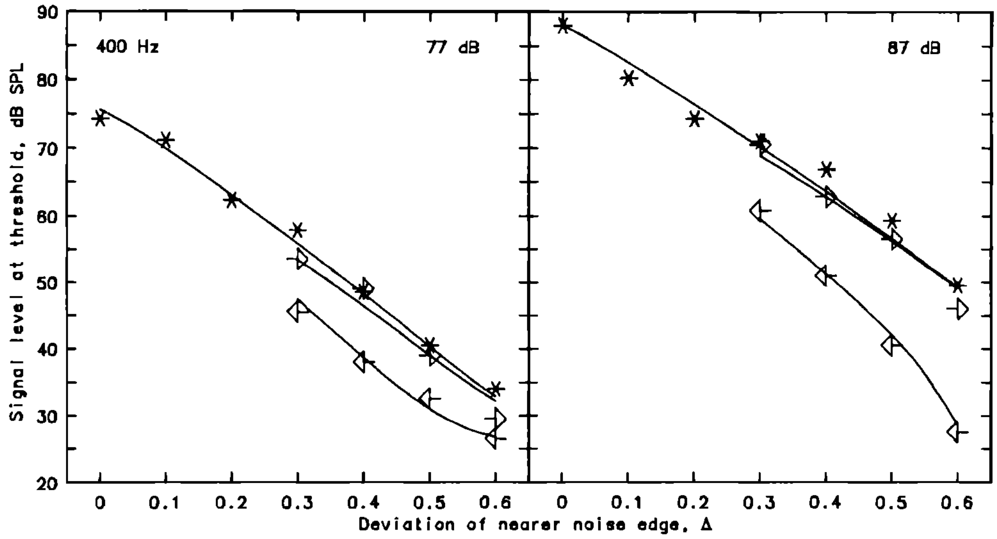


FIG. 4. As in Fig. 1, but showing typical results at $f_s = 400$ Hz.

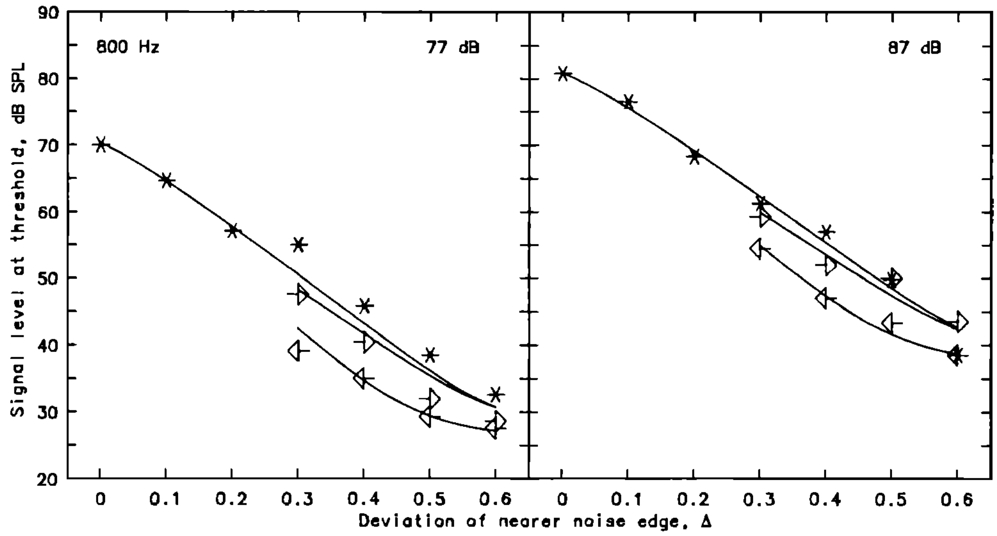


FIG. 5. As in Fig. 1, but showing typical results at $f_s = 800$ Hz.

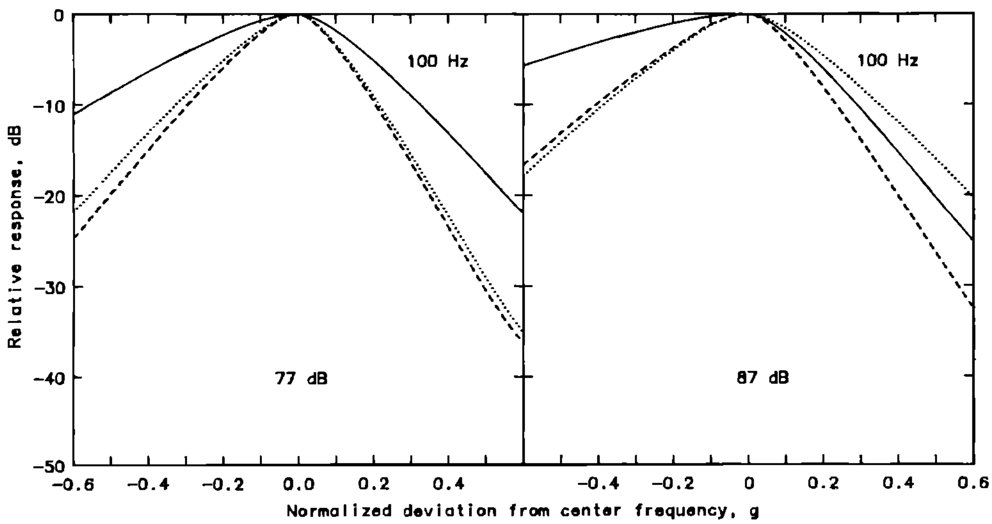


FIG. 6. Each panel shows auditory filters derived from the data of three individual listeners. The three were chosen as having the sharpest (dashed lines), the least sharp (solid lines), and intermediate (dotted lines) filters in terms of their ERBs. The two overall noise levels were 77 dB SPL (left panel) and 87 dB SPL (right panel). The filters were calculated using the ELC + Senn correction. The signal frequency was 100 Hz.

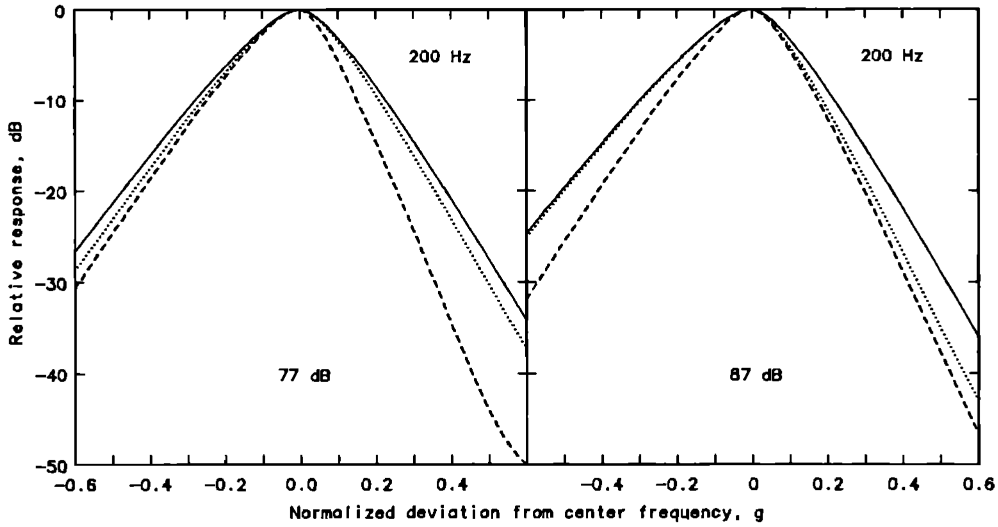


FIG. 7. As in Fig. 6, but for $f_s = 200$ Hz.

800 Hz (Fig. 1 shows typical data sets at $f_s = 200$ Hz). Each figure shows results for one center frequency, with results for the lower and higher noise levels in the left and right panels, respectively. It can be seen that the roex$(p,r)$ model generally provides a good fit to the data, and that discrepancies from the fitted values do not appear to be systematic. The filter shapes are plotted in Figs. 6–9. Table II gives the means and standard deviations of the parameters of the derived filters.

The results are quite consistent across subjects, although the variability tends to increase somewhat at 100 Hz. The standard deviations of the ERBs are about one-tenth of their value for center frequencies of 200, 400, and 800 Hz. This is similar to the standard deviation of the ERB at 2 kHz for young normally hearing subjects (Moore, 1987). The ERBs are reasonably close to the values suggested by Moore and Glasberg (1983), but are somewhat greater at the higher center frequencies. Their Eq. (3) suggests that the ERB should be 38 Hz at 100 Hz, 47 Hz at 200 Hz, 67 Hz at 400 Hz, and 107 Hz at 800 Hz. The mean ERBs obtained, for the 77-dB noise level, were 36, 47, 87, and 146 Hz, respectively. However, the data summarized in Moore and Glasberg (1983) were all obtained using moderate to low noise levels, whereas the present data were obtained with relatively high noise levels. The ERB of the auditory filter increases with increasing noise level (Weber, 1977; Lutfi and Patterson, 1984; Moore and Glasberg, 1987). It is noteworthy that the ERBs obtained by us at the lower two center frequencies are very close to the values suggested by Moore and Glasberg (1983) in spite of our relatively high noise levels. This is consistent with the finding of Rosen and Stock (1989), who used only symmetric notched noises, that the ERB of the auditory filter hardly changes with level at very low center frequencies.

Moore and Glasberg (1987) argued that the ERB values suggested in Moore and Glasberg (1983) were valid for a noise level of approximately 51 dB/ERB; at this noise level, the auditory filter is roughly symmetric, at least for middle frequencies. With increasing noise level, the low-frequency skirt of the filter gets markedly less steep, while the high-frequency skirt may get slightly steeper. The present data are generally consistent with this. The filters are always asymmetric, with shallower low-frequency skirts. The lower skirt is less steep at the higher noise level, for each center frequency. There is, however, no clear trend for the high-frequency

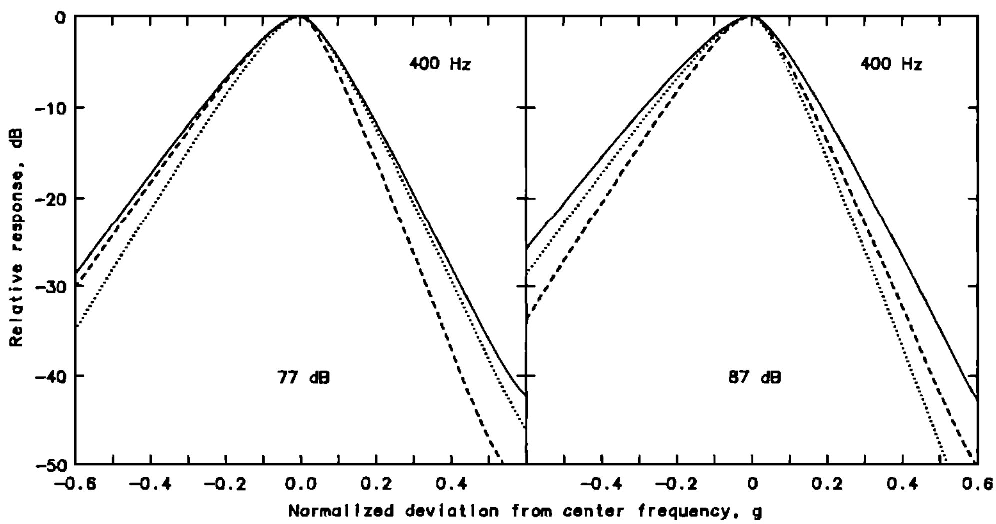


FIG. 8. As in Fig. 6, but for $f_s = 400$ Hz.

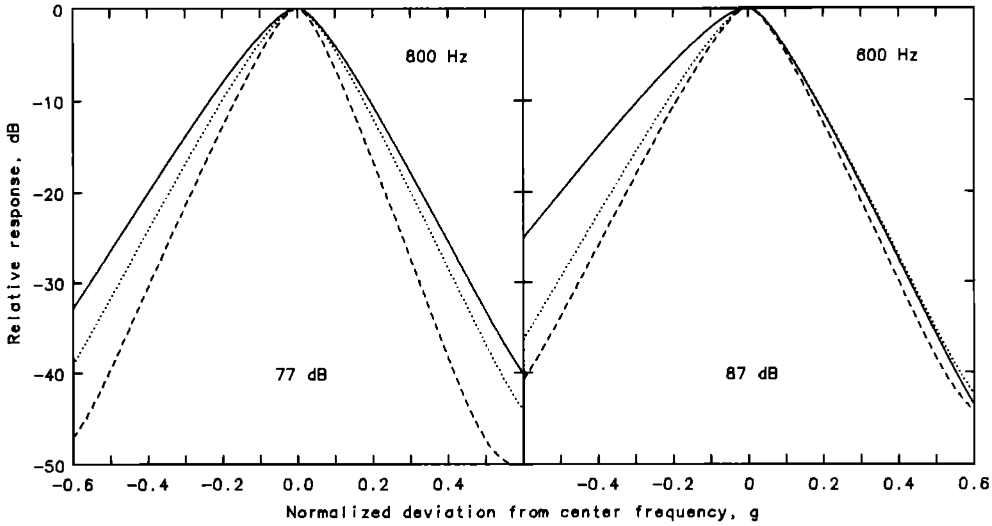


FIG. 9. As in Fig. 6, but for $f_s = 800$ Hz.

skirt to change with level. Data for a wider range of noise levels would be needed to clearly establish whether the high-frequency slope changes with level.

### C. Implications of the bandwidth values

The results clearly show that the ERB of the auditory filter continues to decrease below 500 Hz. This same general conclusion applies whichever correction is used: The Senn correction leads to slightly smaller ERBs, whereas the MAF + Senn correction leads to slightly larger ones. Since most workers would view the ERB of the auditory filter as corresponding to the CB, these results lend weight to the growing body of evidence reviewed in the Introduction, which suggests that the "classical" values of the CB need to be revised.

Data on the ability to "hear out" partials from complex tones also indicate narrow CB values at low center frequencies. Plomp (1964) suggested that a partial could only be heard out from a complex tone if it was separated from neighboring partials by one or more critical bandwidths. However, his data on the ability to hear out partials are discrepant with the classical CB values, indicating rather good resolution of the lower partials even at very low fundamental frequencies. His data are consistent with the narrower values found by us.

### D. The signal-to-masker ratio required for threshold

The values of $K$ in Table II indicate the signal-to-masker ratio at the output of the auditory filter required to achieve threshold; $K$ provides a measure of the efficiency of the detection process following the auditory filter (Patterson and Moore, 1986). It is also closely related to the critical ratio $R$, which is the ratio of the signal power at threshold to the noise spectrum level. If $R$ and $K$ are expressed as power ratios, then $R = K \cdot \text{ERB}$. The values of $K$ in Table II do not show any clear change with overall noise level but increase markedly with decreasing center frequency. This can explain why the critical ratio increases with decreasing frequency below about 400 Hz (Hawkins and Stevens, 1950), whereas the ERB continues to decrease.

There are several possible reasons why $K$ increases at low center frequencies. One is connected with the inherent fluctuations in the noise. At the lowest center frequency, the two noise bands comprising the masker were only 40 Hz wide. Such bands have distinct amplitude fluctuations, both within and across stimuli, and these may impair performance. A second possibility is that, at low frequencies, the signal is harder to detect because fewer cycles occur within the fixed duration of the signal. It should be emphasized that the shape of the auditory filter is inferred from the way that the signal threshold changes with notch width. Factors that

TABLE II. Means and standard deviations (in parentheses) of the parameters of the auditory filters derived from the individual data for each level and center frequency. The filter shapes were obtained using the ELC + Senn correction. The values of $K$ are also given.

| Frequency | Level | $p_l$ | $p_u$ | $r$ | ERB | $K$ (dB) |
|---|---|---|---|---|---|---|
| 100 | 77 | 9.6 (3.0) | 15.5 (3.0) | − 47 (19) | 0.36 (0.10) | 4.3 (1.3) |
| | 87 | 8.5 (2.7) | 12.8 (2.8) | − 60 (19) | 0.42 (0.10) | 3.4 (1.4) |
| 200 | 77 | 14.5 (1.0) | 21.2 (3.2) | − 56 (14) | 0.23 (0.02) | 2.9 (2.7) |
| | 87 | 14.4 (1.3) | 21.3 (2.1) | − 69 (18) | 0.23 (0.02) | 3.0 (1.6) |
| 400 | 77 | 15.6 (1.1) | 22.8 (3.2) | − 55 (7) | 0.22 (0.02) | − 0.2 (2.5) |
| | 87 | 14.8 (1.7) | 24.1 (2.5) | − 66 (18) | 0.22 (0.02) | 1.0 (2.5) |
| 800 | 77 | 20.2 (3.3) | 25.1 (5.0) | − 48 (6) | 0.18 (0.02) | − 2.9 (3.5) |
| | 87 | 16.0 (3.2) | 22.1 (0.7) | − 61 (30) | 0.22 (0.02) | − 3.7 (2.0) |

increase task difficulty for all notch widths should not affect the derived filter shape.

Finally, it is worth noting that the $K$ values reported here are somewhat larger than found in other comparable studies (e.g., Patterson *et al.*, 1982; Fidell *et al.*, 1983; Glasberg and Moore, 1986). However, the earlier studies used a two-alternative forced-choice task targeted on the 71% or 76% correct point on the psychometric function, whereas we used a three-alternative forced-choice task targeted on the 79.4% correct point. This can easily explain the discrepancy.

## IV. CONCLUSIONS

In deriving the shapes of the auditory filters from notched-noise data, account should be taken of the frequency response of the earphone used and of the frequency-dependent attenuation produced by the middle ear. A reasonable approximation to the latter can be obtained by using the shape of the equal-loudness contour for high-level tones. For the relatively high noise levels used in our experiments, the auditory filter was asymmetric, the lower skirt being less steep than the upper skirt. The asymmetry tended to increase at the higher noise level. The mean ERBs of the filters were 36, 47, 87, and 147 Hz for $f_s = 100$, 200, 400, and 800 Hz, respectively. The signal-to-masker ratio at the output of the filter required to achieve threshold increased markedly with decreasing center frequency.

## ACKNOWLEDGMENTS

This work was supported in part by grants from: The Deafness Research Foundation, UNC-CH Medical School Foundation, UNC Faculty Research Council, The Burroughs Wellcome Fund, and the Medical Research Council (U.K.). We thank Michael Stone, Michael Shailer, and Anne Sherman for helpful comments on an earlier version of this paper, Joseph Hall and John Grose for help in generating the noise stimuli and for advice on programming and instrumentation, and Margaret Casteen and Martha Sloan-Clontz for assistance in data collection.

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