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Frequency-dependence of auditory temporal resolution

Kishon-Rabin, Liat, Ph.D.

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300 N. Zeeb Rd.
Ann Arbor, MI 48106

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FREQUENCY-DEPENDENCE OF AUDITORY TEMPORAL RESOLUTION

by

LIAT KISHON-RABIN

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August 28, 1990
Date

Arthur Boothroyd
ARTHUR BOOTHROYD, Ph.D.
Chair of Examining Committee

August 28, 1990
Date

Irving Hochberg
IRVING HOCHBERG, Ph.D.
Executive Officer

IRVING HOCHBERG, Ph.D.

HARRY LEVITT, Ph.D.

Supervisory Committee

MARY FLORENTINE, Ph.D.

External Examiner

The City University of New York

ABSTRACT

Frequency Dependence of Auditory Temporal Resolution

by
Liat Kishon-Rabin

Advisor: Professor Arthur Boothroyd

The purpose of this study was to determine the relationship between auditory temporal resolution and frequency. For this purpose, thresholds for pure tones during a brief gap in an 800 msec white noise burst were measured as a function of gap width and tone frequency in 5 normally hearing subjects. The tone began well before the gap and ended well after, so that any switching transients were masked by the noise.

The results showed strong frequency-dependence below 1 KHz but only weak frequency-dependence above this frequency. The data relating threshold to gap width at a given frequency could not be modeled adequately by a single decay process. The data were, therefore, reanalyzed using a model that included two decay processes plus the "perceptual integration time". This model fits the data very well and can be explained in physiological terms. Specifically:

a) At the onset of the gap, there is persistence of noise within the cochlea due to "ringing" in the auditory filters. This phenomenon determines threshold

for very short gaps.

b) Simultaneously, electro-chemical systems within or beyond the cochlea, recover from threshold adaptation that occurred during the noise. This threshold recovery is measured for longer gaps.

c) The listener detects the tone when its energy has exceeded the higher of the two thresholds determined by a) and b), for some critical time called the "perceptual integration time".

Mean values of the time constants of the assumed cochlear decay process were 13.0, 5.6, 4.0, and 3.6 msec at 0.5, 1, 2, and 4 KHz, respectively. These values show strong frequency dependence and are well predicted from linear auditory filter theory. The mean value of the time constant of the assumed recovery from adaptation was 90 msec and was independent of frequency. Perceptual integration time ranged from 3 to 8 msec and there was weak evidence of frequency dependence.

The findings provide strong support for the notion that peripheral temporal resolution is limited by the time constants of peripheral auditory filters. They also illustrate the benefits of an experimental paradigm that separates peripheral temporal constraints from those of more central mechanisms.

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CHAPTER 1

Introduction

Sound exists in the dimensions of time, frequency, and intensity, and its perception requires a sensory system that can respond to fine detail in these three dimensions. The present study is primarily concerned with the interaction between the first two dimensions, and asks specifically whether the ability of the auditory system to resolve fine temporal detail is dependent on frequency.

The dependence of temporal resolution on frequency is predicted on the basis of a generally held assumption and some well established psychophysical data. The assumption is that the limits of temporal resolution in the human auditory system are peripherally determined. That is, they reflect the cochlea's ability to respond to rapid changes. The psychophysical data are those that show the auditory system to behave as though it contains a large number of overlapping filters whose bandwidths increase with increasing frequency. In mechanical and electrical filters there is a direct trade-off (or inverse relationship) between bandwidth and time resolution. It is, therefore, predicted that temporal resolution in the human auditory system should improve with increasing frequency. Demonstration of such a relationship is one way of providing support for the underlying assumptions, including those implicit in current models of cochlear function.

Attempts to demonstrate the predicted relationship between temporal resolution and frequency have produced mixed results. Some experiments have shown a relationship, others have shown none. Still others have found frequency-dependence for limited ranges of frequency only. Interpretation of these findings, however, is complicated by the fact that several different paradigms have been used. In addition, there are procedural and instrumentational issues that may have confounded results. The experimental data and the procedural issues will be explored in detail in the next chapter.

Psychophysical studies of subjects with sensorineural hearing loss have shown them to have broader auditory filters than do normally-hearing subjects. This observation has led several investigators to test the hypothesis that sensorineural hearing loss leads to improved temporal resolution. Again the results have been mixed, but most studies have found that temporal resolution deteriorates with increasing sensorineural hearing loss rather than improving. As with the studies of normally-hearing subjects, differences of methodology and unresolved procedural complications have made it difficult to interpret the findings.

In the experiment to be described, an attempt has been made to eliminate many of the procedural problems that have been encountered in earlier studies investigating frequency-dependence of temporal resolution. This procedure involves the detection of continuous tones during brief temporal gaps in a white noise stimulus. Although the principle of having tones in gaps has been introduced before, this paradigm is relatively a new one. The expectation was that, if temporal resolution is peripherally determined, high frequency tones will be audible at lower levels than low frequency tones, for a given gap size.

The results of studies of this type are important for several reasons. First they may lead to improved understanding of the normal auditory system. Second, they may lead to improved understanding of the nature of auditory pathology. And third, they may provide insights into appropriate ways of improving auditory perception in the hearing-impaired through the use of hearing aids and other sensory aids.

Chapter 2

Literature Review

Quantitative measures of temporal resolution usually focus on the shortest time interval of perceptual significance. Because perceptual significance can be of several types, there are several different measures of temporal resolution. Some examples are: the ability to determine which of two events is longer; the ability to determine the order of events; and the ability to distinguish one from two events. At first sight it may seem as though these are simply different measures of a single phenomenon. But, in fact, the difference between these measures of temporal resolution may reflect temporal constraints of different stages in the auditory system. The primary focus of the literature to be reviewed here is in measures of temporal resolution that are believed to reflect the early stages at the periphery of the auditory system.

A major problem in investigating temporal resolution in the auditory system is that time and spectrum are not independent. Thus, the detection of a change in time pattern may depend not only on temporal change in time pattern may depend not only on temporal resolution, but also on the detection of differences in spectral content. To avoid the confounding of spectral cues with temporal resolution ability, two general approaches have been used. One approach utilizes signals such as white noise, whose long-term spectra are unlikely to be changed when the time pattern is altered. The spectrum of white noise remains flat, for example, even if the noise is interrupted by a brief gap. The second approach uses stimuli whose spectra are likely to be altered by the change in time pattern, and are therefore introduced with background noise to mask those spectral changes. Both approaches will be discussed in the following review.

Temporal Resolution Measured by the Discrimination of Stimuli with Identical Spectra

1. The Discrimination of Time-Reversed Signals

The long-term spectrum of a sound is not changed when that sound is time-reversed (i.e., played backwards in time). Thus, the ability to discriminate a time reversed signal from the original is assumed to reflect the sensitivity of the listener to the difference in time pattern of the two signals. In 1970, Ronken attempted to measure temporal resolution using time-reversed stimuli. In his study, stimuli were pairs of clicks differing in amplitude. In one pair of clicks the first click was louder than the second. In the second, the order was reversed. Listeners were asked to determine whether the two click-pairs were the same or different. The interval within a click pair was reduced until the click pairs were judged to be the same. The smallest interclick interval for which subjects could reliably perceive the difference in temporal order was found to be 1-2 msec. The discrimination of the click pairs, however, depended on the interclick intensity. For example, discrimination of temporal order was achieved at interclick intervals of 1 - 2 msec when the intensity ratio within each click-pair was between 3 to 10 dB. In examining the results, Ronken was concerned that listeners did not base the discrimination on differences in temporal order, but rather on changes in the quality of the sound produced by forward and backward masking. He hypothesized that when the more intense click preceded the weaker one, there was more masking than in the reverse order, which may have resulted in distinct sounds for each click-pair. To investigate this possibility, forward-backward masking data were obtained from one subject. The results did not support the hypothesis that discrimination between the pairs of clicks was a result of differences in sound qualities due to temporal masking. Thus, Ronken concluded that when using time-reversed stimuli, normally-hearing listeners can detect temporal changes in the order of 1 - 2 msec.

Resnick and Feth (1975) repeated Ronken's experiment and further evaluated the interactive effects of interclick intensity ratio, level of the more intense click and interclick intervals. They used four interclick intensity ratios (3, 6, 12, and 24 dB), two levels of the more intense click (45 and 75 dB SL), and eight interclick interval values (between 0.5 and 10 msec). The duration of each click was 0.1 msec. The results showed that the temporal acuity for clicks ranged from less than 0.5 msec to as high as 1.8 msec. Discriminability of the time-reversed click pairs was found to be dependent on the level of the more intense click and on the interclick intensity ratio. For example, when the interclick intensity ratio was 3 dB, the minimal interclick intervals at which 75% correct discrimination was obtained were 3 msec and 1 msec for 75 dB SL and 45 dB SL clicks, respectively. Thus, it seems that estimates of temporal resolution based on the discriminability of time-reversed click pairs are substantially affected by intensity manipulations. This should, therefore, be taken into consideration when estimating auditory temporal acuity using Ronken's paradigm for both normal and hearing impaired listeners.

Henning and Gaskell (1981) repeated Ronken's (1970) study using very brief clicks (0.02 msec). The more intense click was constant at 30 dB above the level corresponding to 75% correct detection, the interclick intensity ratio values were 1, 2, and 6 dB, and the interclick interval ranged from 0.1 msec to 4 msec. The results showed that two of the three subjects discriminated between the two click pairs when interclick interval was approximately 0.5 msec. Dr. Gaskell, who served as one of the subjects, discriminated correctly in 90% of the trials when the interclick interval was only 0.25 msec. The authors predicted that, given the frequency response of the headphones, there are probably no important differences in the spectra of a 0.25 msec click (as used in Ronken's study in 1970) and a 0.02 msec click, and it is, therefore, unlikely that stimulus duration was the reason for the difference in the results between the two studies. Green (1985) repeated Henning and Gaskell's study but did not achieve better than chance performance for 0.02 msec clicks with a 0.25 msec interclick separation. Thus, the reasons for the discrepancies in the results of the different studies remain unclear. Nevertheless, it appears that discrimination of time-reversed stimuli reflect a time resolution of at least 1 - 2 msec.

Studies with time-reversed stimuli have also been conducted for longer, speechlike sounds (Leshowitz, in Green, 1971). Using a digital filter, noise bursts of 50 msec in duration with equal energy throughout most of the spectrum, were subjected to a pair of filtering operations. The first level of filtering was similar to that imposed by the acoustic resonance of the vocal system. The resulting waveform was essentially a fricative. The second filtering was similar to the first except the impulse response was reversed in time. The intensity patterns within both bursts and the distribution of energy over frequency were the same in both sounds. Thus, distinguishing between the two fricatives on the basis of long-term energy cues was impossible, and discrimination had to be based on differences in the temporal order within the stimuli. The results showed that subjects could discriminate the order of events and reliably distinguish between the "normal" fricative and "time-reversed" fricative when they were separated by only 5 msec.

The conclusion that time-reversed stimuli offer a direct measure of the auditory system's true sensitivity to differences in temporal order, cannot be readily made. Henning and Gaskell offered two alternative explanations to the results. The first explanation suggests that listeners are detecting small changes in the running amplitude spectrum. It has been hypothesized that the auditory system is capable of integrating energy over a brief period of time. Because the long-term amplitude spectra of the stimuli waveforms are identical, the auditory system must integrate information over a short period of time in order to be able to detect those amplitude changes. Thus, the results using time-reversed stimuli may reflect minimal integration time, which in turn, may be an aspect of temporal resolution.

The second explanation suggests that listeners are detecting differences in the duration of stimuli. Papoulis (in Henning and Gaskell, 1981) has shown that, theoretically, duration of the stimulus is related to the amplitude spectrum and the phase spectrum. That is, waveforms with identical amplitude spectra which differ in their phase spectra may also differ in their duration. Specifically, the contribution of a change in phase to duration depends on the value of the power spectrum in the frequency region over which the phase changes take place. If, for example, discontinuities in the phase spectrum occur at a frequency where the power spectrum is zero there would be no effect on the signal's duration. Because the stimuli used in Henning and Gaskell (1981) and Ronken's (1970) studies differed in their phase spectra at frequencies where their power spectra were nonzero, they also differed in their duration. Small and Campbell (1962) found that for interstimulus interval of 600 msec, the just discriminable change in duration for a 0.4 msec noise burst is 0.3 msec. These stimuli are comparable to those used in Henning and Gaskell's study. Thus, it is possible that the discrimination of time-reversed stimuli is cued by duration differences. To test this possibility, Henning and Gaskell attempted to equate both the spectra and the durations of the two click pairs. This objective, however, could not be accomplished due to equipment limitations: the corresponding changes in the temporal waveforms were small relative to the quantization error. They suggested that this issue may be resolved with 15-bit quantization. Thus, the possibility that listeners are discriminating differences that are functions of phase differences but do not require sensitivity to phase differences as such, cannot be precluded.

In summary, estimates of temporal resolution in the range of 0.5 to 5 msec, have been obtained, using the discrimination of time-reversed pair-click stimuli method. It is not clear, however, to what extent these values reflect the true temporal resolution of the auditory system. It is possible that the results were confounded by other factors such as duration cues and should, therefore, be interpreted cautiously.

2. The Detection of Gaps in Broadband Noise

As mentioned previously, the long term spectrum of broadband noise remains the same if the noise is briefly interrupted. Plomp (1964), therefore, suggested that the threshold for gap detection in a broadband noise may provide a simple and convenient measure of temporal resolution. In his study, using a two-alternative forced-choice paradigm, two normally-hearing subjects were asked to determine whether a silent gap was in the first or the second interval. The study was repeated at 15 sensation levels (SLs) ranging between 5 and 75 dB SL in 5 dB steps. The results showed the minimal detectable gap to be approximately 3 msec for noise levels greater than 25 dB SL. At sensation levels lower than 25 dB SL, gap detection thresholds increased, i.e., became worse. Plomp also investigated the effects of the sensation of each of the two maskers surrounding the gap on the minimal detectable gap. Gap thresholds were obtained when the first masker was presented at each of three sensation levels: 25, 45, and 65 dB, and the second masker at each of 12 sensation levels ranging between 10 and 65 dB in 5 dB steps. The results showed that gap detection threshold (msec) increased linearly with decreasing thresholds of the second masker (dB) when time was plotted logarithmically. Furthermore, regardless of the level of the first masker, the data seem to converge to one point at zero sensation level of the second noise burst at approximately 200 msec.

Plomp hypothesized that after the first noise masker was turned off, the sensation of the noise continued to persist but with a gradual decay. Only after the sensation decayed beyond some critical point, was the auditory system capable of detecting the gap. The minimal detectable gap is assumed to reflect the period of time required for the noise to decay to some critical value after which the onset of the second noise masker is detected. For any gap value smaller than the minimal detectable gap, the auditory system would perceive the two maskers as one. Plomp also hypothesized that the rate of decay of the auditory sensation can be estimated by reducing the intensity of the second masker and obtaining gap detection thresholds accordingly. The results showed that the noise in the auditory system after it has been turned off decays linearly as a function of log time. Thus, the auditory sensation of the noise would decrease much faster for smaller time values than for longer ones.

Penner (1977) investigated the effects of the relative durations of the leading and trailing noises on gap detection. Gap thresholds were determined when the duration of the leading noise was fixed at 200 msec and the trailing noise ranged randomly between 200 to 400 msec and vice versa. That is, the duration of the trailing noise was fixed at 200 msec and the duration of the leading noise varied randomly between 200 to 400 msec. The results showed that, when the duration of the leading noise was randomized, gap thresholds were constant at 2 - 3 msec. When the duration of the trailing burst was randomized, gap thresholds increased only when the difference in intensity levels between the leading and trailing noises were large and gap durations were greater than 30 msec. Note that randomizing the duration of either noise resulted in randomization of the total duration. Because the results showed that randomization of the first noise did not affect gap threshold, it was therefore assumed that the total duration of the stimulus was not a cue. Under certain conditions, however, the duration of the trailing noise is relevant to gap detection.

In summary, results of gap detection studies suggest that when the surrounding noise level is greater than 25 dB SL, it takes approximately 3 msec for the sensation of the internal noise to decay to some critical value after which another stimulus can be detected. The results are affected by the level of the surrounding noise bands and the duration of the trailing noise. This method, as described here, does not provide frequency specific information. Furthermore, it is not clear to what extent the minimal detectable gap reflects true temporal resolution or other properties of the auditory system such as intensity resolution. This will be discussed in greater detail in the following sections.

3. Detection of Amplitude Modulation

The experiments described above each give a single number to describe temporal resolution. A more general approach is to measure the threshold for detecting changes in the amplitude of a sound as a function of the rapidity of the changes. Rodenburg (1977) and Viemeister (1977) were the first to pursue this approach. In their studies, white noise was sinusoidally modulated, and the threshold for detecting the modulation was determined as a function of the modulation rate. The function relating threshold to modulation rate was referred to as the temporal modulation transfer function (TMTF). The results showed that threshold was independent of modulation rate for rates up to 16 Hz and then threshold increased with increasing frequency up to 1.0 KHz. At modulation rates above 1.0 KHz the modulation could not be detected at all. Thus, the resulting TMTF resembled a low-pass filter with a 3 dB down point of about 50 Hz. It was also found that the shape of the TMTFs did not vary much with overall level. The ability to detect the modulation, however, became worse at low sensation levels. In general, these results suggest that for low modulation rates performance was limited by the amplitude resolution of the ear. As the rate increased beyond 16 Hz, temporal resolution started to have an effect and modulation rates greater than 1.0 KHz were too rapid for the ear to follow. Thus, the auditory system seems to become progressively less sensitive to amplitude changes as the rate of modulation increases.

The Effects of Center Frequency on Temporal Resolution as Measured by the Discrimination of Stimuli with Identical and Non-Identical Spectra

The studies described thus far used broadband stimuli primarily to ensure that any changes in the time domain would not provide spectral cues. Therefore, they provide no information as to whether the temporal resolution of the ear varies with stimulus' frequency. It has been suggested that, in theory, temporal resolution may be poorer at low frequencies than at high. This suggestion is based on several hypotheses. The first is that the peripheral auditory system behaves as if it contains an array of bandpass filters which analyze a complex signal into its spectral components (Fletcher, 1940). It is known that the duration of the temporal response of a physical linear filter to an abrupt cessation of an input, varies inversely with filter bandwidth: the narrower the bandwidth of the filter the longer its response time.

Moreover, the time constant of the impulse response at the output of the filter is halved for each doubling of frequency. There is evidence to show that the bandwidth of the auditory filter increases with increasing frequency (e.g. Moore and Glasberg, 1981; Glasberg, Moore and Nimmo-Smith, 1984). A simple analogy with the physical filter would suggest a trade-off between auditory filter bandwidth and temporal resolution. Thus, if the responses of the auditory filters limit temporal resolution, temporal resolution is expected to be worse at low frequencies than at high. Furthermore, the time constant of the auditory filter should decrease by a factor of two for each doubling of frequency. The experiments described in this section test this hypotheses.

1. Time-Reversed Sinusoids

Green (1973) used time-reversed stimuli similar to those used by Ronken (1970) to measure temporal resolution. In order to obtain frequency specific information, he used brief pulses of a sinusoid of 1.0, 2.0, and 4.0 KHz. Each stimulus consisted of a brief pulse of a sinusoid in which the level of one half of the pulse was 10 dB different from the other half. Thus, all stimuli had the same long-term power spectrum but differed in the order of events in time i.e., in phase. The ability to discriminate between two such stimuli was measured as a function of stimulus duration. Testing was conducted both with and without a background noise masker to further reduce possible cues from spectral splatter. The duration of the stimuli which corresponded to 75% correct discrimination was similar for all frequencies and ranged between 2 - 4 msec. It appeared, however, that for the very short stimulus durations (2 - 6 msec), listeners based their decisions on differences in quality between the two signals as reported in Ronken's (1970) experiment. Also, no differences were found between the masked and unmasked conditions.

In summary, when using time-reversed sinusoids, temporal resolution is estimated to be 3 msec independent of frequency. Note, however, that the results are confounded by the fact that for short stimuli, subjects based their decisions on quality differences. Thus, it is possible that the results reported here did not reflect true temporal resolution but rather, some other process.

2. <u>The Discrimination of Huffman Sequences</u>

Patterson and Green (1970) and Green (1973) acknowledged the possibility that the discrimination of time-reversed stimuli may have been based on quality differences rather than on differences in the time domain. They studied a different class of signals which have the same long-term spectrum, but differ in their short-term spectra. These sounds are called Huffman sequences. These are brief broadband sounds where the energy in a single frequency region is delayed relative to that in other regions. The amount of the delay, the center frequency of the delayed frequency region, and the width of the delayed frequency region can all be varied. The ability to discriminate between stimuli of a pair of Huffman sequences implies that the ear is sensitive to the differences in the arrival time of energies at different frequencies. It is assumed, therefore, that the discrimination of Huffman sequences reflects the temporal resolution of the auditory system.

In generating Huffman Sequences, the degree of tuning (center frequency/bandwidth) of a digital all-pass filter is varied. In general, an all-pass filter has a flat amplitude response across all frequencies. It is composed of overlapping high-pass and low-pass sections where the phase response can vary. This region of phase change can be considered the "center frequency" of the all-pass filter. A sharply tuned filter changes phase rapidly and a large delay is imposed on the energy at this frequency. Similarly, if the filter has a relatively broad bandwidth, the energy in the center frequency is delayed by a small amount. The results of Green's (1973) study showed that 90 dB SPL stimuli were correctly discriminated when energy delays were 1.86, 2.26, and 1.99 msec at center frequencies of 0.65, 1.9, and 4.2 KHz, respectively. Similar results were reported by Jesteadt, Bilger, Green and Patterson (1976). They tested three subjects with energy delays in frequency regions centered at 0.472, 1.024, 1.968, and 4.016 KHz. When Huffman sequences were presented at 90 dBSPL the minimum discriminable energy delays were 1.5 to 2 msec. These results suggest that when using Huffman sequences, temporal resolution is independent of frequency.

Several possibilities have been offered to explain the lack of frequency dependence in normally-hearing listeners using Huffman sequences. Patterson and Green (1970) suggested that the source of limitation of temporal acuity as being measured in this task may be more central than the first-stage filtering process.

Shailer and Moore (1983) suggested that in discriminating Huffman sequences or time-reversed sinusoids, listeners performed a two-interval forced-choice task by detecting any difference between the waveforms on the two halves of a trial. In principle, this could be achieved even if the impulse response of the auditory filter was very prolonged, provided the listener was able to store a sufficiently accurate record of the output of the filter. Buus and Florentine (1985) suggested that because listeners were presented with a pair of Huffman sequences where the energy was delayed at a different center frequency in each stimulus, at least two different auditory channels were stimulated. The task, therefore, may be measuring the timing of events between channels tuned to different frequencies and not temporal resolution at a particular frequency. Thus, it could not be concluded that the discrimination of Huffman sequences provides a direct measure of temporal acuity.

3. Detection of Gaps in Narrowband noise

Another approach for investigating frequency dependence of temporal resolution has focused on gap detection tasks in narrow-band noise. Boothroyd (1973) obtained gap detection thresholds using low-pass filtered stimuli as part of a study that tested the effect of high frequency information on gap detection threshold. Four normally-hearing listeners were tested in four conditions: no filtering (i.e., broadband noise), and low-pass filtering at 2000 Hz, 1000 Hz and 500 Hz. Minimum gap detection values were found to be 3.89, 5.48, 6.06, and 7.62 msec, respectively. It can be seen that gap detection threshold improved as the cutoff frequency increased. Note, however, that the attenuation rate of the filter was relatively gradual (approximately 24 dB/oct.). Thus, spectral energy at the off-band frequency regions may have been audible and, therefore, provided cues from higher frequencies. Furthermore, as the cutoff frequency increased, the bandwidth of the stimulus increased as well. Recent studies have shown that when the upper cutoff frequency of the stimulus was fixed, gap detection threshold improved as the lower cutoff frequency was lowered, thus making the bandwidth of the stimulus wider (Shailer and Moore, 1985; Grose, Eddins and Hall, 1989). These results suggest that listeners are able to combine information across channels. It is not clear, however, to what extent Boothroyd's results reflect changing bandwidth rather than changing center-frequency (or upper frequency boundary).

As mentioned earlier, one of the problems resulting from the introduction of a brief gap in a narrowband noise is the production of spectral components across a broad frequency band. This may render the gap audible, not as a gap, but as a click. To ensure that listeners would not hear any unwanted spectral components, Fitzgibbons and Wightman (1982) used low-level background noise whose spectrum resembled that of the "splatter" associated with the gap in the signal, but it had a sharp notch at the tested frequency. This masker is also referred to as a notched-noise masker. The low-pass and high-pass cutoffs of the notch were set, respectively, to match the low and high frequency limits of the bandpass stimulus. The attenuation rates of the filters producing the notch were 96 dB/oct. This paradigm ensured that listening was restricted to limited spectral regions. Fitzgibbons and Wightman tested 5 normally-hearing listeners with three octave-band stimuli: 0.4-0.8, 0.8-1.6, and 2.0-4.0 KHz. The test stimuli were presented at 85 and 65 dB SPL corresponding to 50 and 30 dB SL (re: masked threshold). Mean gap-detection thresholds for stimuli presented at 85 dB SPL were 9.17, 6.97 and 5.09 msec, whereas, the gap means at 65 dB SPL were 12.38, 9.46 and 6.06 msec (for octave-bands of 0.4-0.8, 0.8-1.6, and 2.0-4.0 KHz, respectively). It can be seen that the gap thresholds improved with increasing cutoff frequencies of the bandpass as predicted by a linear filter theory. Note also that gap detection thresholds obtained for the 2-4 KHz stimuli were approximately 2 msec poorer than those reported for broadband stimuli.

Somewhat similar results were reported by Florentine and Buus (1983). They measured the minimal detectable gap in an octave band of noise presented with its complementary notched-noise masker as a function of frequency and level in three normally-hearing listeners. The results suggested a strong frequency effect through 8.0 KHz. For example, at 83 dB SPL, the minimal detectable gap decreased from approximately 20 msec at 0.5 KHz, to 9 msec at 2.0 KHz, and to 6.0 msec at 4.0 KHz. These results are consistent with the general prediction that temporal resolution improves with increasing frequency. Note, however, that the absolute gap detection values reported by Florentine and Buus are higher than those reported by Fitzgibbons and Wightman (1982). It is possible that procedural differences such as stimulus bandwidth, signal-to-masker ratio, and bandwidth of the masker's spectral notch had an affect on the results.

The effect of stimulus bandwidth on gap detection thresholds was further explored by Fitzgibbons (1983). In one series of measurements, gap detection thresholds were compared under progressive degrees of high-pass and low-pass filtering. The second series of measurements were taken with narrow-band signals centered at octave intervals between 0.6 and 4.8 KHz with bandwidths of 0.25, 0.5, and 1.0 KHz. All stimuli were presented with notched-noise maskers. Three normally-hearing subjects were tested using the Bekesy tracking procedure where the duration of the gap was fixed and the subject tracked the minimum stimulus intensity to keep the gap at threshold. The results showed that gap detection thresholds remained constant at 3.76 msec for the high-pass filtered stimuli regardless of the cutoff frequency. For the low-pass filtered stimuli, gap detection thresholds improved from 11 msec for a cutoff frequency of 0.5 KHz to 3.76 msec for a cutoff frequency of 6.0 KHz. For the narrow band stimuli, gap detection threshold improved with increasing center frequency from 9.0 msec at a center frequency of 0.6 KHz to 2.67 msec at a center frequency of 4.8 KHz. Also, when the signal bandwidth increased from 2.0-4.0 KHz to 0.4-4.0 KHz (note that the upper cutoff frequency was kept constant), gap detection threshold did not improve. From these results, Fitzgibbons concluded that listeners use information from the highest frequencies available to them and that the bandwidth of the stimulus does not affect gap thresholds.

A month after Fitzgibbons (1983) reported his results, Shailer and Moore (1983) reported on a study in which they tested the effects of variations in masker's and signal's bandwidth on gap detection thresholds. Bandpass stimuli centered at nine frequencies ranging from 0.4 to 8.0 KHz were used. The spectral width of the signal was half of the center frequency and the notch width was 54 % of the center frequency. Shailer and Moore chose the bandwidths to be a constant proportion of center frequency because the auditory filter bandwidth is generally agreed to be roughly a constant proportion of center frequency. They argued that maintaining a constant relative bandwidth ensured that the signal-to-background ratio was roughly constant at the output of the auditory filter at the center frequency. Thus, the spectral notches used in this study were narrower than the octaveband notched maskers used by Fitzgibbons and Wightman (1982). By narrowing the spectral notch of the masker, the frequency range available to subjects for each tested frequency was further limited. Using an adaptive, two-interval, forced-choice procedure, mean gap detection thresholds of three listeners improved monotonically with frequency. Thresholds were 22.5 msec at 0.2 KHz, 8.0 msec at 1 KHz, 4.25 at 4 KHz, and 3.2 msec at 8.0 KHz. To ensure that the variation of gap thresholds with frequency did not stem from the variation in bandwidth with center frequency, a second experiment was conducted in which the effects of bandwidth on thresholds of gap detection were investigated. Gap thresholds were found to decrease with increasing signal bandwidth at all center frequencies, in roughly an exponential fashion. For example, for center frequency of 1.0 KHz, gap detection thresholds were 12.5 msec and 6.9 msec for 0.25 KHz and 1.0 KHz bandwidths, respectively.

There are at least three factors that possibly contributed to the improved gap detection thresholds with increasing bandwidth in Shailer and Moore's (1983) study. These include: a) the signal-to-masker ratio (SMR) in the stimulus, b) the inherent fluctuations in the noise, and c) the bandwidth of tested signal. These will be discussed in turn.

a) <u>The effect of signal-to masker ratio</u>

In Shailer and Moore's study, the depth of the spectral notch in the masker decreased with decreasing notch width, because of the limited slopes of the filter used (96 dB/oct). At the narrowest signal/masker bandwidth, for example, the spectrum level at the minimum in the notch was only 11 dB below the spectrum level in the masker passband. Thus, the gap was not silent, but was partially filled in by the background noise, which could have impaired performance. Further support to this explanation was provided by Shailer and Moore (1985). They measured gap thresholds as a function of signal-to-masker ratio (SMR) at center frequencies 0.5, 1.0 and 6.5 KHz with three signal bandwidths at SMR of 0, 3, 6, 9, 12, 15, and 20 dB. Higher values of SMR were not tested because pilot studies indicated that at SMR of 25 dB, clicks associated with the gap were heard. The results showed that for center frequencies 0.5 and 1.0 KHz, gap detection thresholds improved with increasing SMR. At 6.5 KHz, gap thresholds improved significantly from 0 to 9 dB, after which, only slight improvement was observed. For all frequencies, at a given SMR value, gap thresholds improved with increasing signal's bandwidth. This effect was more pronounced at the low frequencies than the high. These results suggest that there is a strong relationship between the auditory filter's bandwidth, the signal's bandwidth, and the SMR. It was suggested that as the signal bandwidth became close to, or less than, the bandwidth of the auditory filter, the SMR at the output of the auditory filter would be considerably less than the nominal SMR, and thus gap thresholds may be influenced by SMR even for ratios exceeding 15 dB. This issue was further investigated by Green and Forrest (1989). They measured gap detection thresholds at two center frequencies, 1.0 and 2.0 KHz, with different signal bandwidths, notch bandwidths (of the masker) and SMRs. The results showed that increase in notch bandwidth did not affect gap threshold as long as the notch was equal to or wider than the signal bandwidth. Furthermore, gap thresholds significantly increased (i.e., became worse) with decreasing of SMR for signal bandwidths equal to, or less than, the bandwidth of the filter centered at the tested frequency. For example, gap thresholds for 1.0 KHz varied between 6 and 35 msec depending on the signal bandwidth, notch bandwidth and the SMR. Note also that the results may have been influenced by the overall level of the stimuli. Fitzgibbons (1984) showed that the SMR required for detection of a given gap decreases slightly with overall level.

In summary, introduction of a notched-noise masker to a narrowband stimulus allows one to obtain frequency specific information on gap detection. In this procedure, spectral information that may provide the listeners with cues for gap detection are being masked. Results of studies, however, show that gap thresholds improve as the bandwidth of the spectral notch and the SMR increase. Also, the effects of the bandwidth of the notched-noise masker and the SMR vary with filter bandwidth. Thus, gap detection thresholds in passband stimuli with background noise appear to reflect a confounding of temporal resolution with the effects of the masker.

b) The effect of fluctuations in the noise

The second explanation for the better performance at broader bandwidths relates to the inherent random fluctuations in intensity in the noise signal. The rapidity of these fluctuations increases with increasing bandwidth. When both the noise bandwidth and the auditory-filter bandwidth are large, the fluctuations in the noise at the output of the auditory filter are rapid, and of low amplitude, and are therefore not likely to be confused with the gap. When the noise bandwidth is narrow, or when the noise is centered at a low-frequency where the auditory-filter bandwidth is narrow, the fluctuations at the output of the filter are large and therefore more confusable with the gap. Thus, for very narrow noise bandwidths, gap thresholds are hypothesized to increase (become worse) and change little with center frequency. This hypothesis was supported by the results of DeFillipo and Black Snell (1986) who measured gap detection thresholds in five normally-hearing listeners using very narrow band noise at center frequencies 0.25, 0.5 and 1.0 KHz. Gap thresholds were found to increase from 8.5 msec to 14.6 with decreasing frequency. As expected, these thresholds were higher than previously reported and varied very little with center frequency (approximately 2 msec/octave). Similar results were reported by Shailer and Moore (1985) who found only slight gap threshold changes as a function of frequency.

In summary, in order to obtain temporal resolution measures in very limited frequency regions, the bandwidth of the tested signal must be reduced. This, in turn, introduces problems related to the inherent fluctuations in the noise amplitude, which, in turn, may be affected by the bandwidth of the auditory filter. Again, the results do not appear to reflect only temporal resolution of the auditory system.

c) <u>The effect of signal's bandwidth</u>

The third explanation for the improved thresholds with increasing bandwidth found in the study by Shailer and Moore (1983) is that listeners may be using information from the highest frequency region available. To test this hypothesis, Shailer and Moore (1983) tested two subjects with two different signal bandwidths. The upper cutoff of the signal band and masker notch were held constant while the bandwidth of both was doubled by extending the lower cutoff frequency. The results showed that when the bandwidth of the signal, whose upper cutoff frequency was held constant at 1.0 KHz, increased from 0.5 KHz to 1.0 KHz, gap thresholds remained relatively unchanged (7.73 msec and 7.65 msec, respectively). These results are consistent with those reported by Fitzgibbons (1983). In summarizing their results, Shailer and Moore concluded that subjects restrict their listening to the highest frequency available and that they cannot combine information across auditory filters very effectively in gap detection tasks.

The effects of signal bandwidth on gap detection thresholds were further investigated by Shailer and Moore in 1985. Gap detection thresholds were obtained by three subjects for stimuli whose upper cutoff frequency was constant at 0.425 KHz and the bandwidth varied between 50 and 400 Hz. For comparison, gap detection thresholds were also obtained for stimuli with constant center frequency and bandwidth which varied between 0.0625 and 1.0 as a proportion of center frequency. Contrary to previous results (Fitzgibbons, 1983; Shailer and Moore, 1983), the results of this study showed that for stimuli with fixed upper cutoff frequency, gap detection thresholds decreased (i.e., improved) with widening of bandwidth even when the bandwidth exceeded the auditory filter bandwidth at that frequency. Gap thresholds decreased from 29.3 msec to 16.8 msec when the bandwidth increased from 100 to 400 Hz, even though the auditory filter's bandwidth centered at 0.425 is less than 100 Hz (Moore and Glasberg, 1983). The variation of gap threshold with bandwidth, however, was found to be more gradual for fixed high-frequency cutoff than for fixed center frequency. The results support the hypothesis that listeners use information from the higher frequency regions. The fact that thresholds improved with increasing bandwidth even when the upper cutoff frequency was constant suggests that listeners had some limited ability to combine information from different auditory filters centered at the lower frequencies.

These results were further supported by Grose, Eddins and Hall (1989). They measured gap thresholds in stimuli with two fixed high-frequency cutoffs at 0.6 and 2.2 KHz but with varying low-frequency cutoffs. Stimulus bandwidth varied from 25 to 600 Hz and from 50 to 1600 Hz for the lower and higher high cutoff frequencies, respectively. The results showed that gap thresholds improved with increasing bandwidth. Furthermore, at a given bandwidth, gap thresholds were similar for both frequencies. These results support the hypothesis that, in terms of gap detection, the auditory system <u>is</u> capable of combining information across the outputs of multiple auditory filters. Moreover, gap detection may not be limited by the output of a particular filter.

It can be concluded that measured gap detection thresholds are greatly influenced by the auditory filter centered at the highest frequency in the stimulus. However, listeners seem to be able to combine information for gap detection from lower frequencies as well. It is important, therefore, that, when investigating frequency effects of temporal resolution with gap detection in bandpass noise, both upper cutoff frequency and bandwidth should be considered.

In summary, gap detection in bandpass noise offers a way to investigate frequency dependence of temporal resolution. To avoid any spectral cues caused by the introduction of the gap, the stimuli are supplemented with background noise with a spectral notch at the center frequency of the test signal. Results of studies using this type of stimulus to estimate the frequency effects have shown that gap detection thresholds improve with increasing frequency, thus in keeping with the auditory filter theory. Note, however, that temporal resolution can be predicted by a linear filter theory for low frequencies only. (e.g. Shailer and Moore, 1983). A summary of these results are shown in Table 2.1. These estimates of temporal resolution have shown to be confounded with procedural factors, such as, the signal bandwidth, the upper cutoff frequency of the signal, the bandwidth of the masker's notch, and the signal-to-masker ratio all affect gap threshold. Moreover, the magnitude of their effects varies as a function of frequency. A most striking feature of the work just summarized is that a lot of time has been spent on resolving procedural issues relating to the possibility of artifacts and to the confounding of several phenomena without a clear picture emerging on frequency dependence of temporal resolution.

Table 2.1

Summary of Minimal Gap Detection Studies in Normally-Hearing Listeners. Data are Presented in Msec as a Function of Frequency. Intensity Levels of Stimuli Ranged between 60 to 85 dBSPL.

		Center Frequency (KHz)					
Authors	Stimulus	<0.5	0.5	1.0	2.0	4.0	>4.0
Boothroyd (1973)	LP	--	7.6	6.1	5.5	--	3.9
Fitzgibbons & Wightman (1982)	BP+NNM	--	9.2	7.0	--	5.1	--
Fitzgibbons (1983)	HP+NNM	--	3.5	3.5	3.5	3.5	3.5
	LP+NNM	--	11.0	8.8	6.8	4.7	3.8
	BP*+NNM	--	8.2	6.0	4.0	--	2.7
Shailer & Moore (1983)	BP*+NNM[1] BP**+NNM	22.5	--	4.3	--	4.3	3.2
Shailer & Moore (1985)	BP*+NNM1	29.2	12.0	8.0	--	--	4.0
Buus & Florentine (1985)	LP*+ NNM[2]	40.0	20.0	12.0	9.0	6.0	4.0
	LP & HP	--	--	--	6.0	--	3.0
DeFillipo & Black Snell (1986)	BP**	14.6	10.2	8.5	--	--	--
Glasberg, Moore & Bacon (1987)	BP*+NNM	--	9.0	7.0	6.0	--	--
Formby & Muir (1988)	HP	--	--	--	--	--	2.9
	LP	--	--	7.6	5.4	4.1	--
Moore & Glasberg (1988)	BP*+NNM	--	9.0	6.8	6.1	--	--
	Sine+NNM	--	3.3	3.7	4.2	--	--
Green & Forrest (1989)	BP*+NNM[1]	--	--	13.0	6.0	--	--
	BP*+NNM[2]	--	--	19.0	11.0	--	--

Key is on next page

Key for Table 2.1

BP = Bandpass noise
HP = high-pass filtered noise
LP = low-pass filtered noise
NNM^1 = Notch noise masker presented at a S/N of 5 dB or more.
NNM^2 = Notch noise masker presented at a S/N of 0 dB.
* = bandwidth increased as the center frequency of the passband increased.
** = constant passband bandwidth.

4. Detection of Gaps in Sinusoids

The problems associated with narrowband noise stimuli in gap detection studies led researchers to offer yet another method which uses sinusoidal stimuli to obtain gap thresholds at specific frequencies (Shailer and Moore, 1987, Moore and Glasberg, 1988; Green and Forrest, 1989; Moore, Glasberg, Donaldson, McPherson and Plack, 1989). To mask the spectral "splatter" produced as the result of the onset and offset of the gap in the stimulus, the sinusoids were presented in continuous noise with a spectral notch at the tested frequency. Generally, results of the cited studies were found to be strongly affected by the phase at which the signal was turned off and on (to produce the gap). Shailer and Moore (1987) reported that when the sinusoid started at the phase it would have had if it continued without interruption (preserved phase), detection of the gap in percent correct increased with increasing gap size. For conditions when the sinusoid started at positive-going zero crossing or negative-going zero crossing, performance (in percent correct) increased non-monotonically with increasing gap size. These results were explained in terms of the ringing of the auditory filter. Since the gap is partially filled due to the ringing of the auditory filter, the phase of the sinusoid at the beginning and end of the gap would either further fill the gap thus making the dip at the output of the filter more difficult to detect or, increase the depth of the gap and therefore make it easier to detect. Using the preserved phase condition, Moore and Glasberg (1989) measured gap threshold in sinusoidal stimuli of different frequencies. The results showed similar gap thresholds of 5 msec at frequencies 0.4, 1.0 and 2.0 KHz. Only at 0.2 KHz was the gap threshold was slightly higher (7 msec).

Somewhat similar results were reported by Green and Forrest (1989). They found gap thresholds of 11 and 9 msec at 0.5 and 1.0 KHz, respectively, and a constant gap threshold of 6 msec at 2.0 and 4.0 KHz. These results were surprising since it was expected that the ringing in the auditory filter would lead to larger gap thresholds at low center frequencies. Moore et al (1989) argued that, in listening to gaps in sinusoidal stimuli, listeners are highly dependent on their ability to detect dips at the output of their auditory filter. The decision as to whether or not there was a gap would, therefore, depend on both intensity resolution and on temporal resolution. This hypothesis was supported with studies with the hearing impaired and is further discussed in the next section.

In summary, the use of sinusoidal stimuli in gap detection studies provide means to obtain frequency specific information without introducing inherent random fluctuations of stimulus' amplitude that can be confused with the gap. There are, however, several disadvantages. These include the need to use notch-noise maskers to mask unwanted spectral components (the effects of the notched-noise masker on gap thresholds have been reviewed in Section 3 of this chapter), the fact that listeners may depend primarily on his/her intensity resolution in addition to temporal resolution for detecting the gap, and the dependency on the phase of the signal at the onset and offset of the gap for gap detection. Thus, it is important that these issues be considered when examining and interpreting results of gap detection in sinusoidal stimuli.

Temporal Acuity in Sensorineural Hearing Loss of Cochlear Origin

The literature reviewed so far discussed temporal resolution in normally-hearing listeners. For these listeners, temporal resolution is expected to improve with increasing frequency (e.g., Fitzgibbons and Wightman, 1982; Shailer and Moore, 1983; Buus and Florentine, 1985) because the bandwidth of the auditory filter is assumed to increase with increasing frequency (e.g. Moore and Glasberg, 1981; Glasberg, Moore and Nimmo-Smith, 1984). Of indirect relevance to the issue of frequency dependence of temporal resolution are results of studies with cochlear-impaired listeners. For these listeners, the auditory filters have been found to be wider than those of normally hearing at the same frequency (e.g. Marcellino, 1977; Tyler, 1986; Glasberg and Moore, 1986). If temporal resolution is indeed a direct consequence of auditory filter bandwidth, it is expected to improve for subjects with sensorineural hearing loss. Contrary to expectations, studies using gap detection tasks reported larger gap detection thresholds in noise stimuli for listeners with cochlear impairment than for normally-hearing listeners. This was found to be true whether the stimulus used to mark the gaps was broadband noise (e.g. Irwin, Hinchcliff, and Kemp, 1981), low-pass noise (e.g. Boothroyd, 1973; Florentine and Buus, 1984) or bandpass noise presented in a broadband or a notched- noise background (e.g. Fitzgibbons and Wightman, 1982; Irwin and Purdy, 1982; Tyler et al, 1982; Glasberg, Moore, and Bacon, 1987). It is evident, therefore, that some factor other than the impulse response of the hypothetical auditory filter limits the hearing-impaired listeners performance. It is possible that this "unknown" factor is responsible for the increase in gap thresholds with decreasing center frequency.

In making comparisons between normally hearing and hearing-impaired listeners, at least three important factors must be considered. These include: a) the effective frequency range available to the listener, b) the presentation level of the stimulus (SPL vs SL), and c) the bandwidth of the stimulus. These factors will be discussed in turn.

a. <u>The effect of the audible frequency range</u>

There is considerable evidence that, for normally-hearing listeners, thresholds for the detection of gaps in bandlimited noise decrease with increasing center frequency and with increasing bandwidth (Fitzgibbons and Wightman, 1982; Fitzgibbons, 1983; Shailer and Moore, 1983, 1985; Buus and Florentine, 1985; De Fillipo and Black Snell, 1986; Glasberg, Moore, and Bacon, 1987; Moore and Glasberg, 1988; Green and Forrest, 1989; Grose, Eddins and Hall, 1989). For broadband stimuli, it appears that listeners primarily use information from the highest frequency region available (e.g. Shailer and Moore, 1983; 1985). Thus, for listeners with high-frequency hearing losses, performance may be poorer because the higher frequency components in the stimuli are inaudible. This would decrease both the effective bandwidth and the effective upper cutoff frequency as was true for most of the subjects in the studies by Fitzgibbons and Wightman (1982), Tyler et al (1982), Florentine and Buus (1984), and Buus and Florentine (1985).

b. The effect of presentation level

The second important factor to consider when making comparisons between normal and impaired hearing is the level at which subjects are tested. Gap thresholds decrease with increasing level both for normal and for hearing-impaired listeners (Boothroyd, 1973; Fitzgibbons and Wightman, 1982; Shailer and Moore, 1983; Florentine and Buus, 1984; Buus and Florentine, 1985; Glasberg, Moore and Bacon, 1987;

Moore and Glasberg, 1988). It remains unclear whether impaired and normally hearing listeners should be compared at equal sound pressure levels (SPLs), equal sensation levels (SLs), or some other level, such as equal loudness. Boothroyd (1973) and Fitzgibbons and Wightman (1982) found that hearing-impaired listeners had larger gap thresholds than normally-hearing listeners regardless of whether the comparison was made at equal SPL or equal SL. The difference was, however, considerably smaller when stimuli were presented at equal SL. Tyler et al (1982) found that, at approximately equal SL, hearing impaired listeners had larger gap thresholds than normal at 4.0 KHz but not at 0.5 KHz.

Florentine and Buus (1984) attempted to separate the effects of hearing-impaired listeners' elevated thresholds from the effects of other alterations in their impaired auditory system by comparing gap detection thresholds of hearing-impaired listeners and normally-hearing listeners who had simulated hearing loss. The simulation of hearing loss was achieved by presenting a spectrally shaped masker whose spectrum mimicked the audiogram of the hearing impaired subjects. They found that only three of the seven hearing-impaired listeners showed gap thresholds consistently larger than those found for the listeners with simulated impairments. Florentine and Buus concluded that for the majority of the hearing-impaired listeners, the results could be accounted for entirely by elevated pure-tone thresholds. There are, however, some hearing impaired-listeners whose increased gap detection thresholds cannot be accounted for by presentation level. For these listeners, it is possible that reduced temporal resolution is caused by an increased time constant of another processing level in the auditory system, which follows the filtering process. Humes, Espinoza-Varas and Watson (1988) attempted to model sensorineural hearing loss, using normally hearing subjects with simulated hearing loss. According to the authors, the model provided a good description of the data from studies of frequency resolution it did not predict accurately the results of studies measuring temporal resolution. They concluded, therefore, that for some hearing impaired subjects for whom reduced sensitivity cannot explain their worse-than-normal temporal acuity, temporal resolution is indeed abnormal.

Further investigations on the effect of presentation level on gap detection thresholds in the hearing impaired were conducted by Fitzgibbons and

Gordon-Salant (1987). They tested the minimum SLs required for gap detection in 5 listeners with different degrees of sensorineural hearing loss. They found that the measured SL required for gap detection thresholds were weakly correlated ($\underline{r}$ = -0.47) with subjects' pure-tone thresholds. This finding was inconsistent with the data of normally-hearing listeners with simulated hearing loss who exhibited a strong correlation ($\underline{r}$ = -0.97) between threshold elevation induced by masking and the minimum SL required for gap threshold. Furthermore, no significant correlation was found between gap threshold and the degree of hearing loss. In fact, the two listeners who exhibited close to normal hearing sensitivity produced the largest gap thresholds. Unlike some of Florentine and Buus' (1984) findings, Fitzgibbons and Gordon-Salant could not directly attribute the enlarged gap thresholds to the listeners' configuration of hearing loss. The results did not indicate a predictable relationship between the degree of hearing loss in the cochlear-impaired listeners and the minimum sensation levels required for gap detection thresholds. It was therefore concluded that the supra-threshold intensity cues for gap detection in cochlear-impaired listeners need not be the same as those used by normally-hearing listeners with simulated hearing loss. Thus, the results of the hearing impaired cannot be simply explained by their elevated pure-tone thresholds.

Glasberg, Moore and Bacon (1987) attempted to provide further information on the degree to which subjects with cochlear hearing impairment show deficits in gap detection over and above that which might be expected on the basis of the elevation of pure-tone thresholds. They measured gap detection thresholds of 17 hearing-impaired listeners, 9 of whom had unilateral losses. Each of these nine listeners served as his/her own control, the normal ear being compared to the impaired ear at both equal SL and SPL. The results showed that gap thresholds for the impaired ear were usually, but not always, larger when the comparison was made at equal SPL and for four of the nine subjects when stimuli were presented at equal SL. Glasberg and his colleagues concluded that although the differences in SL of the stimuli could account for some of the differences between normal and impaired ears, there was a substantial source of variation that could not be explained in this way. Note also that Glasberg et al found better-than-normal gap thresholds for a few of the hearing-impaired listeners. It has been suggested that for these listeners, the reduced (better) gap thresholds may be related to the broadening of their auditory filters.

Glasberg et al (1987) tested a few possible explanations, other than reduced threshold of audibility, for the elevated gap thresholds of the hearing-impaired listeners. First, they examined the possible link between gap detection and forward masking since both are assumed to depend on the rate of recovery from adaptation. Specifically, they wanted to test the hypothesis that poor gap detection in impaired ears might be associated with slower-than-normal rate of recovery from forward masking. Thresholds of audibility for three pure-tones in a forward masking paradigm were measured in 5 listeners with unilateral cochlear hearing loss, 2 with bilateral hearing loss, and 2 normally hearing, all of whom had participated in the gap detection study. The results showed that there was a relationship between the rate of recovery from masking and gap detection thresholds but not a strong one ($\underline{r} = -0.41$, $\underline{p} = 0.05$). Furthermore, both gap thresholds and the values measuring the rate of recovery from masking were correlated with absolute thresholds ($\underline{r} = 0.64$, $\underline{p} < 0.01$ and $\underline{r} = -0.68$, $\underline{p} < 0.01$, respectively). After partialing out the effect of absolute threshold, the correlation of the gap thresholds with the rate for recovery from masking became close to zero (r = 0.04, p > 0.05). Thus, Glasberg et al could not rule out the possibility that both the gap thresholds and recovery from forward masking are mediated by their relationship to absolute threshold.

Glasberg's et al (1987) second possible explanation for the poorer-than-expected gap detection thresholds in the hearing impaired is in a longer than normal time constant of the temporal integrator. The temporal integrator is a hypothetical device used to model the ability of the auditory system to integrate energy over time. It has been suggested that, when a stimulus is presented, the output of the temporal integrator rises with time following an exponential function (e.g. Plomp and Bouman, 1959; Zwislocki, 1960). The parameter of this exponential function is called the time constant of the device. If during the presentation of the stimulus a certain integration level is reached, that stimulus is detected and the level is considered as threshold for detection. More details on the temporal integrator will be found in the last section of this chapter.

To test the possibility that the time constant of the temporal integrator in the hearing impaired is longer than that in the normal hearing, Glasberg et al (1987) obtained a measure of temporal integration from the difference in absolute threshold for long tones (210 msec) and short tones (10 msec). The average threshold differences for the hearing-impaired listeners were consistently shorter than for the normally hearing. It may be, however, that the temporal integrator involved with absolute threshold is not the same as that involved with gap detection since the time constants of the two appear to be different (de Boer, 1985; Green, 1985). Nevertheless, there does not appear to be evidence to support Florentine and Buus' (1984) hypothesis that the large gap thresholds found for some hearing-impaired listeners can be explained by an increase in the time constant of the temporal integrator.

Another possible explanation for the poor gap detection thresholds in the hearing-impaired offered by Glasberg et al (1987) is the magnitude of the detectable change at the output of the hypothetical temporal integrator. The smaller that change, the smaller the gap threshold. Based on this hypothesis, gap thresholds are expected to correlate with measures of intensity discrimination. Glasberg et al examined the relationship between the intensity DL for 210 msec pulsed-tones of different frequencies and gap thresholds in each impaired ear. The correlations failed to reach statistical significance with one exception. Gap thresholds at 1.0 KHz were significantly correlated with the intensity DL for pulse tones (r < 0.48, p < 0.05). This correlation increased slightly after the effect of absolute threshold was partialed out (r = 0.53, p <0.05). Thus, there was some weak evidence that differences in the sensitivity of the detector mechanism following the hypothetical temporal integrator may have influenced gap thresholds. Moreover, these results suggest that gap detection thresholds need not be a direct result of temporal resolution but rather of the intensity resolution of the auditory system.

b. The effect of stimulus' bandwidth

The bandwidth of the stimulus is another important factor to consider when making comparisons between gap detection thresholds of normal and hearing-impaired listeners. As mentioned previously, gap detection in normals improves with increasing stimulus bandwidth. It has also been pointed out that the inherent fluctuations in the stimulus' envelope become perceptually more pronounced for narrower bandwidths. The imposed gap would, therefore, be more likely to be confused with on-going fluctuations in the stimulus for narrow bands of noise than for wider bands (e.g. Shailer and Moore, 1983; 1985; 1987; Green, 1985; Glasberg et al, 1987; Grose, Eddins and Hall, 1989). Fluctuations in the noise may be of particular importance in hearing-impaired listeners due to the presence of loudness recruitment. It is expected that the fluctuations in loudness associated with intensity fluctuations in the noise would be greater for these listeners, making dips in the noise sound more like gaps. Although both normal and impaired listeners have reported hearing a gap in both intervals of a forced choice trial, this phenomenon appeared to be more marked for listeners with cochlear impairments (Glasberg et al, 1987; Moore and Glasberg, 1988; Grose, Eddins and Hall, 1989).

The role of the inherent fluctuations in noise on gap detection threshold in hearing-impaired listeners was examined by Moore and Glasberg (1988). They compared gap thresholds measured with noise bands to those measured with sinusoidal stimuli, assuming the latter have no inherent fluctuations. Gap detection thresholds were measured in seven subjects with moderate unilateral hearing loss. The impaired ear was compared to the normal ear for each subject at equal SPL and SL. The bandwidths of the noise band stimuli increased with increasing frequency. Both noise bands and sinusoidal stimuli were presented in a notched-noise masker. The results showed that for the noise stimuli, gap thresholds were generally larger for the impaired ears than for the normal ears when the comparison was made at equal SPL (21.4, 12.6, 9.4 msec versus 9.0, 6.8, 6.1 msec for 0.5, 1.0 and 2.0 KHz, respectively), but the difference was reduced when the comparison was made at equal SL (15.3, 12.0 and 7.1 msec for normal ears at 0.5, 1.0, and 2.0 KHz, respectively). For the sinusoidal markers, gap thresholds for the impaired ears were not generally larger than those for the normal ears when tested at equal SPL, except at 0.5 KHz (8.1, 4.7, 4.3 msec versus 3.3, 3.7 and 4.2 msec for frequencies 0.5, 1.0 and 2.0 KHz, respectively). When the comparison was made at equal SL, gap thresholds were often found to be smaller for the impaired ear than for the normal ear (7.3, 10.3 and 6.8 msec for normal ears at 0.5, 1.0, and 2.0 KHz, respectively).

To further test the hypothesis that the differences in the results between sinusoidal and noise stimuli were due to the inherent fluctuations in the noise, Moore and Glasberg correlated the dynamic range (as the measure of loudness recruitment) with gap detection thresholds. Gap thresholds were found to be negatively correlated with dynamic ranges ($r = -0.36$, $p < 0.05$ for a one-tailed test). However, both dynamic ranges and gap thresholds were each found to be correlated with absolute threshold ($r = -0.62$, $p < 0.01$ and $r = 0.46$, $p < 0.05$, respectively). When the effect of absolute threshold was partialed out, the correlation between dynamic range and gap thresholds was reduced significantly ($r = -0.1$, $p > 0.05$). Thus, the association between gap threshold and dynamic range could have been due to the association of both with absolute threshold. The results, therefore, did not support unequivocally the hypothesis that fluctuations in the noise are more confused with gaps in the hearing-impaired listeners due to loudness recruitment (or reduced dynamic range).

Moore, Glasberg, Donaldson, McPherson and Plack (1989) found that at equal SL, gap thresholds in sinusoidal stimuli tend to be smaller in the hearing impaired than in normally-hearing listeners. They hypothesized that gap detection thresholds depend on the listeners' sensitivity to dips in the temporal excitation pattern. In other words, they suggested that gap detection thresholds in sinusoidal stimuli depend on intensity resolution. Indeed, the better performance for the hearing impaired at low SLs may have been the reflection of the fact that intensity discrimination is generally better for impaired than for normal ears at low SLs (Glasberg and Moore, 1989). Moore et al (1989) concluded, therefore, that gap thresholds for sinusoids may not provide a direct measure of temporal resolution since they are strongly influenced by intensity resolution.

The better-than-normal gap thresholds in sinusoids in the hearing impaired listeners at equal SL are in agreement with data reported by Jesteadt, Bilger, Green, and Patterson (1976). They measured the discrimination of Huffman sequences in 10 listeners with sensorineural hearing loss. For eight of the listeners, temporal acuity was found to be better-than-normal. No significant correlations were found between the measure of temporal acuity and temporal integration, speech discrimination scores, presence or absence of recruitment, the slope of the pure-tone audiogram, or the etiology of the hearing loss. Note that because no frequency dependence has been observed with Huffman sequences (as reported previously), it has been suggested that this measure of temporal resolution reflects a more central process than the first-stage filtering process. Therefore, the comparison between results of a gap detection task, which possibly reflects a peripheral process, and discrimination of Huffman sequences may be inappropriate.

In summary, listeners with cochlear impaired hearing loss usually demonstrate poorer gap detection thresholds than normally hearing listeners. These results do not support a simple auditory filter model of gap detection. Possible factors that have contributed to the discrepancies between the reported data and the predictions from the model are the reduced available frequency range in the hearing-impaired, the low sensation level and the bandwidth of the stimuli. There are data to suggest, however, that even if these factors were to be partialed out, there would be some hearing-impaired listeners for whom the increased gap thresholds cannot be accounted for. In general, the improved gap thresholds with increasing frequency in the hearing impaired listeners may be caused by broadening of the auditory filters. However, the overall increase in gap detection threshold in the hearing impaired (in comparison to normally hearing), seem to imply some damage to their temporal mechanisms which probably offsets any effects of the broadened auditory filter.

Temporal Acuity in Cochlear Implant Recipients

Temporal acuity in subjects with cochlear implants is of particular interest since they have no frequency selectivity and their auditory filters are assumed to be absent. Thus, any deficits in temporal mechanisms caused by cochlear damage should be maximized. Furthermore, apical and basal regions of the cochlea can be directly stimulated without being affected by the mechanics of the basilar membrane. Assuming that gap detection is indeed limited by cochlear processes, implant subjects would be expected to demonstrate better-than-normal gap detection thresholds because their performance would not be limited by "ringing" of the auditory filters.

Moore and Glasberg (1988) measured gap detection thresholds in noiseband and sinusoidal stimuli in three subjects with single-channel extracochlear implants. They found better-than-normal gap detection thresholds in sinusoidal stimuli when presented at levels just below the discomfort level. For example, one subject achieved gap thresholds of 2 and 1 msec for 0.5 and 1.0 KHz markers. These results were better than those obtained with narrow band noise stimuli and can probably be explained in terms of the confusability of the gap with dips in the noise. Another finding was the increased gap thresholds in narrow band noise stimuli increasing frequency. This can probably be attributed to the reduced dynamic range at high frequencies.

Shannon (1989) measured gap detection thresholds as a function of stimulus level and position of the electrode in the cochlea in 17 subjects with multielectrode, intracochlear implants (Nucleus and Symbion implants). Stimuli were sinusoidal and biphasic pulsatile electrical waveforms for the Symbion and Nucleus implant subjects, respectively. The results showed that gap thresholds were strongly dependent on stimulus level, decreasing from 20 - 50 msec near quiet threshold to 1 - 5 msec for loud stimuli. These results are similar to the range and magnitude of gap thresholds in normally-hearing listeners. Also, no difference was observed in gap thresholds for stimulation at the apical or basal end of the electrode array. This result was consistent with previous findings (e.g. Green, 1973; Moore and Glasberg, 1988) that showed no inherent difference in gap detection in the central auditory system when using information from apical or basal regions of the cochlea. Shannon concluded that since the subjects have total cochlear dysfunction, temporal resolution as measured by the ability to detect gaps is not impaired by cochlear damage.

In summary, the data thus far support the prediction that subjects with cochlear implants exhibit normal or better-than-normal gap detection threshold. Moreover, these results support the hypothesis that gap detection in normally-hearing subjects is limited by the mechanics of the basilar membrane.

Modelling Temporal Resolution

In the previous sections the role of the auditory filter in limiting temporal resolution was considered. While there was evidence that the auditory filter does play a role in some measures of temporal resolution, its influence was seen mainly at the lower frequencies (below 1 - 2 KHz). The response of the auditory filter at the high frequencies is too fast to be a limiting factor in most temporal resolution tasks. This has led to the hypothesis that beyond the auditory filter, at a higher level in the auditory system, there may be other processes which limit temporal resolution.

Buus and Florentine (1985) adapted and simplified Duifhuis' (1973) model of forward masking to propose a model for gap detection. Their detailed description of the model includes input-output relations of an acoustic stimulus containing a gap at four levels in the auditory periphery. At the first level, the acoustic stimulus is assumed to pass through a bank of bandpass filters which represent the frequency analysis along the basilar membrane. The output waveforms from these filters depend on the mechanics of the basilar membrane. Specifically, Buus and Florentine assumed that the ringing of the filter has an exponential decay (e.g., Moore and Glasberg, 1981) with a time constant equal to the inverse of the critical bandwidth. Therefore, low-frequency stimuli would ring longer at the output of the filter than high-frequency stimuli. Results of studies showed that this stage has a significant influence at the low frequencies where gap thresholds became significantly worse with decreasing frequency. Moreover, Shailer and Moore (1983) found a very high correlation ($\underline{r}$ = 0.995) between gap threshold and auditory filter bandwidth for two subjects for frequencies 0.4 and 1.0 KHz. For these frequencies, gap thresholds were inversely proportional to frequency. At frequencies greater than 1.0 KHz, gap thresholds continued to improve but less than predicted by a linear filter theory. Thus, some other limiting factor is assumed to exist for the higher frequencies.

At the second level of Buus and Florentine's model, the output waveform of the initial bandpass filter is either squared or rectified (half-wave). This stage is a reflection of neurophysiological events at the basilar membrane. Specifically, it has been shown that the neural spikes tend to occur for the rarefaction phase of the signal of the stimulating waveform. That is, only deflection of the hairs on the outer hair cells towards the outside of the cochlea, produced by movement of the basilar membrane towards the tectorial membrane, leads to neural excitation. No excitation occurs when the basilar membrane moves in the opposite direction. Thus, the output of this nonlinear device is always positive.

At the third level, the output of the rectifier is fed into an integrator. This integrator (also called a temporal integrator), sums the input energy over a fixed period of time. By doing so, it smoothes the representation of the stimulus so that rapid temporal changes are lost and slower ones are preserved. The time interval over which the energy is summed is referred to as the temporal "window". This window is assumed to slide in time, so that the output of the temporal integrator is like a running average of the input. The temporal window is considered to be a weighted function and has been mathematically modelled by an exponential function (Munson, 1947; Zwislocki, 1969) giving more weight to power at the end of the window than at times close to the beginning of the window. Some experimental data have shown that the temporal window is indeed asymmetric (Moore et al. 1988; Plack and Moore, 1990) and its shape does not vary with center frequency (although the effects of the auditory filter can produce an apparent broadening of the window at low frequencies; Plack and Moore, 1989). Other data suggested that the temporal window is frequency dependent (Penner, Robinson and Green, 1972; Penner and Cudahy, 1973).

At the fourth level, the output of the temporal integrator is then fed into a decision device. The decision rule associated with this model suggests that the gap will be detected if the dip in the output of the temporal integrator exceeds a certain threshold value. It has been suggested that for normally-hearing listeners this threshold value is in the order of 1 - 2 dB (Moore et al, 1989). Thus, temporal resolution seems to be strongly dependent on the difference limen for intensity (Plomp, 1964; Moore et al, 1989).

In summary, Buus and Florentine's model of gap detection assumes that in the process of transforming the acoustic stimulus into its neural representation at the periphery of the auditory system, the stimulus passes through a filter, a non linear device, a sliding temporal integrator, and a decision device. The ability to detect the gap in the stimulus is, therefore, dependent on: the shape and the bandwidth of the initial bandpass filter, the type of the nonlinearity assumed, the type of integrator, the shape of the weighting function describing the temporal window, and the nature and sensitivity of the device following the integrator. Moreover, it is clear from this model that gap detection thresholds may reflect time constants of either one of the proposed processing level or some combination of them. Note that this model assumes that a simple transformation of the stimulus magnitude is integrated over time. Furthermore, it gives no account of the non-linear phenomena observed in the auditory periphery (e.g., combination tones).

Other, more complicated models assume that neural activity rather than stimulus magnitude is integrated over time. This model assumes that the neural spikes evoked by a stimulus is not directly proportional to the stimulus intensity. Rather, it is a nonlinear, compressive function of stimulus intensity i.e., the number of spikes grow more slowly than the stimulus intensity. Several investigators have, therefore, suggested models of temporal resolution in which there is a compressive nonlinearity prior to the temporal integrator (Penner, 1980; Penner and Shiffrin, 1980). In these models, temporal resolution is also determined by the type of nonlinearity assumed, as well as by the temporal integrator and the sensitivity to changes in the output of the integrator.

The models described are conceptually similar. At present there appears to be no strong reasons for preferring one approach over the other. Note, however, that all models account for tasks which require resolution within a particular frequency channel. Thus, tasks which require comparisons across *different* frequency channels, such as, discrimination of Huffman sequences, cannot be described by any one of the current models. Models for this type of task are yet to be offered.

Forward Masking

A different paradigm used in relation to investigations of temporal resolution is forward masking. In this method, threshold for detection of a test signal is measured as a function of its separation from the termination of a preceding stimulus (masker). Because the measure of forward masking is expressed in decibels, it does not provide a single value of temporal resolution as does, for example, the gap detection task. For this reason, this has not been a common method used in investigations of temporal resolution.

Theoretically, gap detection and forward masking can be assumed to have similar underlying processes because both the detection of the gap and the detection of the test signal depend on the decay of the internal noise, possibly due to ringing, to some critical value. Indeed, studies of forward masking have shown the threshold of the test stimulus to decrease as its separation from the end of the masker increases (e.g. Zwislocki, Pirodda, and Rubin, 1959; Weber and Moore, 1981; Jesteadt, Bacon and Lehman, 1982). However, studies investigating the relationship between gap detection and forward masking have reported conflicting results.

Glasberg, Moore and Bacon (1987) investigated the relationship between forward masking and gap detection in cochlear-impaired listeners and found a weak correlation ($\underline{r}$ = -0.41, $\underline{p}$ = 0.05) between the rate of recovery from masking and minimal gap detection. Furthermore, they found that both measures were highly correlated to absolute threshold. When partialing out this latter effect, the correlation of gap thresholds with rate of recovery from masking was close to zero ($\underline{r}$ = 0.04). Glasberg et al concluded, therefore, that the two paradigms reflect different underlying processes. Furthermore, they hypothesized that temporal resolution is dependent on intensity resolution, whereas the threshold of the test stimulus in forward masking is dependent on recovery from masking.

In contrast, Smiarowski and Carhart (1975) investigated the relationship between forward masking and gap detection and found the results to be similar. To make the two paradigms more comparable, level of the second noise burst in the gap detection task was adjusted for the gap to be just detectable. Gap durations ranged from 5 to 80 msec. The results for the two paradigms showed a similar two-stage time course of decay. The first stage was a brief one lasting approximately 3 msec. In the following stage the decay was slower and lasted longer (approximately 75 msec). Thus, Smiarowski and Carhart concluded that when using comparable procedures, forward masking and gap detection reflect the same underlying processes.

There are several important factors to consider when interpreting data from forward masking studies. One such factor is the use of external filters. Any band pass filter used to restrict the frequency range of the stimulus or the masker has its own time constants that will limit the rate of change of output level. There is always the danger that filter time constants will become confounded with physiological time constants.

Another factor which may have an effect in studies of forward masking is the ability of the subject to use "off-time" listening. The term "off-time" listening has been coined by Moore et al (1988) to refer to a process in which the subject bases perceptual decisions on information whose origin is outside the time frame of the physical stimulus. To be more specific - just as a masking noise can be hypothesized to persist, at a physiological level, beyond its physical termination, so can a probe tone. Thus, evidence for the presence of the tone is available after the tone is finished and may be used by the perceiver. The term "off-time" listening was chosen to parallel "off-frequency" listening when subjects are asked to detect tones in band-limited noise (Patterson and Nimmo-Smith, 1980).

Finally, another factor to consider in studies of forward masking is the effect of temporal integration. Because the task in this paradigm is to detect a signal, it is not clear whether temporal summation plays an important role in this detection process or not. Physiological studies have shown that the auditory system responds to the change of neural firing rate. In forward masking, for example, the auditory system will detect the transition from spontaneous neural discharge to high discharge such as that occurring at the onset of the maskee (e.g., Abbas, 1979; Harris and Dallos, 1979). Moreover, behavioral studies have failed to show an effect of the maskee's duration on thresholds for detection (Penner, 1977, Smiarowski and Carhart, 1975). These last results, however, may have been confounded by other factors such as, "off-time" listening. Thus, the issue of the extent of the effect of temporal integration on the detection of the maskee in a forward masking paradigm has yet to be determined.

In attempting to apply forward masking data to Buus and Florentine's (1985) model it is not apparent which level of processing it reflects. That is, it may reflect the first stage filtering process, the time constant of the temporal integrator, a combination of both or some other process.

In summary, the forward masking paradigm may provide insight to temporal resolution of the auditory system. It is not clear, however, which level of processing the data actually reflect. Also, results from forward masking may be confounded by factors such as external filter response and "off-time" listening.

General Summary

It has been suggested that temporal resolution is a function of the time constants of hypothetical filters at the auditory periphery. This hypothesis is based on a simple analogy with electrical filters in which there is a trade-off between filter bandwidth and temporal resolution. Thus, for normally-hearing subjects, when tested at different frequencies, it has been predicted that temporal resolution will improve with increasing frequency because of the increased bandwidth of the auditory filter at the higher frequencies (e.g. Moore and Glasberg, 1981; Glasberg, Moore and Nimmo-Smith, 1984). For hearing-impaired subjects, it has been hypothesized that temporal resolution will be better than in normally-hearing subjects because of the widening of the auditory filters believed to accompany sensorineural damage (e.g. Marcellino, 1977; Tyler, 1986; Glasberg and Moore, 1986).

Different paradigms and different tasks have been used to measure temporal resolution. These include discrimination of Ronken stimuli, discrimination of Huffman sequences, detection of amplitude envelope modulation, gap detection, and forward masking. Using these measures in normally-hearing subjects, the results have not supported, unequivocally, the hypothesis that temporal resolution improves with increasing frequency. Some were in keeping with this prediction (Boothroyd, 1973; Fitzgibbons and Wightman, 1982; Buus & Florentine, 1985) but others were not (Patterson & Green, 1970; Green, 1973; Jesteadt, Bilger, Green & Patterson, 1976; Moore, Glasberg, Plack & Biswas, 1988). Still other studies have shown only partial frequency dependence as predicted by a linear filter theory (e.g. Shailer & Moore, 1983). Thus, in normally hearing listeners, the results in general, are inconclusive and do not support a linear auditory filter theory of temporal resolution.

Results of studies using hearing-impaired subjects have also reported conflicting results. Contrary to expectations, the majority of the results have shown that temporal resolution in the hearing-impaired subjects is poorer than in normally-hearing subjects (e.g. Boothroyd, 1973; Irwin, Hinchcliff and Kemp, 1981; Fitzgibbons and Wightman, 1982; Tyler, Summerfield, Wood and Fernandes, 1982; Florentine and Buus, 1984; Glasberg, Moore and Bacon, 1987). There have been, however, cases where temporal resolution in the hearing impaired were reported to be better than in the normally hearing (e.g. Jesteadt, Bilger, Green and Patterson, 1976; Glasberg et al, 1987).

It appears, therefore, that filter broadening with increasing frequency in normally-hearing subjects is not equivalent to auditory filter broadening in subjects with sensorineural hearing loss. In normally-hearing subjects, temporal resolution either remains the same or improves with increasing frequency. In hearing-impaired subjects, however, temporal resolution either remains the same as in normals or becomes worse.

There are a few possible explanations for the conflicting results in the published studies measuring frequency dependence of temporal resolution. One explanation may be related to the different paradigms used. It is possible that the various tasks reflect different levels of processes in the auditory system. Some of these processes may be dependent on peripheral filtering effects while others may be more dependent on central factors.

Another possible explanation for the conflicting results may be related to the procedural problems associated with the experiments investigating frequency dependence of temporal resolution. These problems include: a) time constants of external filters used to limit frequency range of stimuli and with which the time constants of the auditory system may be confounded; b) spectral "splatter" ,i.e., the production of spectral components across a broad frequency band when a brief gap is introduced in a continuous noise, thus, rendering the gap audible as a click rather than a gap; c) inherent fluctuations in the noise that may be confused with the gap to be detected (Shailer and Moore, 1983, 1985, 1987; Green, 1985; Glasberg et al 1987); and d) "off-time" listening which allows the subject to use the output of a single temporal window that gives the highest signal-to-masker ratio, but is not centered at the temporal center of the signal (Moore et al, 1988). This last effect may also allow the listener to integrate information over a period that includes "off-time" or "post-probe" listening.It is not clear to what extent these problems, or the methods used to overcome them, may have influenced the results. It may be, therefore, that a procedure that does not involve external filters, switching of frequency specific stimuli, gap detection in band-limited noise, and "off-time" listening, would provide a more appropriate test of the hypothesis that temporal resolution in the normal hearing will improve with increasing frequency. Such a method was devised for the study to be described.

CHAPTER 3

PURPOSE, RATIONALE and METHODOLOGY

Purpose and Rationale

The purpose of the present study was to investigate the relationship between temporal resolution and frequency in the normal auditory system using a new paradigm. In this paradigm, threshold of audibility for pure-tones during a temporal gap in a white noise stimulus was measured as a function of frequency and gap duration. The onset and offset of the tone occurred during the noise markers. This paradigm was designed to avoid some of the procedural problems mentioned in relation to the published studies, specifically, spectral splatter, external filters, "off-time" listening, and inherent random fluctuations in the noise. Spectral splatter was avoided by using continuous tones and switching only white noise; frequency specificity was obtained by the use of pure-tones rather than by filtering noise; "off-time" listening was avoided by constraining subjects to listen within a temporal gap in the noise; and, random fluctuations in the noise could not be mistaken for target stimuli because subjects were not required to detect the gaps.

The rationale underlying this method is illustrated in Figures 3.1 and 3.2 and was based on the following hypotheses:

(1) When a gap is introduced in white noise, the sensation of the noise does not cease immediately but decays gradually over time (Figure 3.1).

(2) The rate of decay of sensation increases with increasing frequency (Plomp, 1964). This differential rate of decay of sensation is illustrated in Figures 3.1 and 3.2.

(3) If a tone is introduced in a gap, its threshold is determined by the level of the tone in relation to the decaying sensation of the noise. That is, the tone becomes audible when the decaying sensation of the internal noise at surrounding frequencies fall below the level of the tone for a sufficient length of time. For that gap, threshold of audibility of the tone in relation to its threshold in noise is the index of the "depth" of the "perceptual notch". This index can be used to indicate the frequency dependence of temporal resolution as shown in Figure 3.2.

(4) The threshold of the tone at different gap widths traces the decay of sensation of the internal noise in the auditory system.

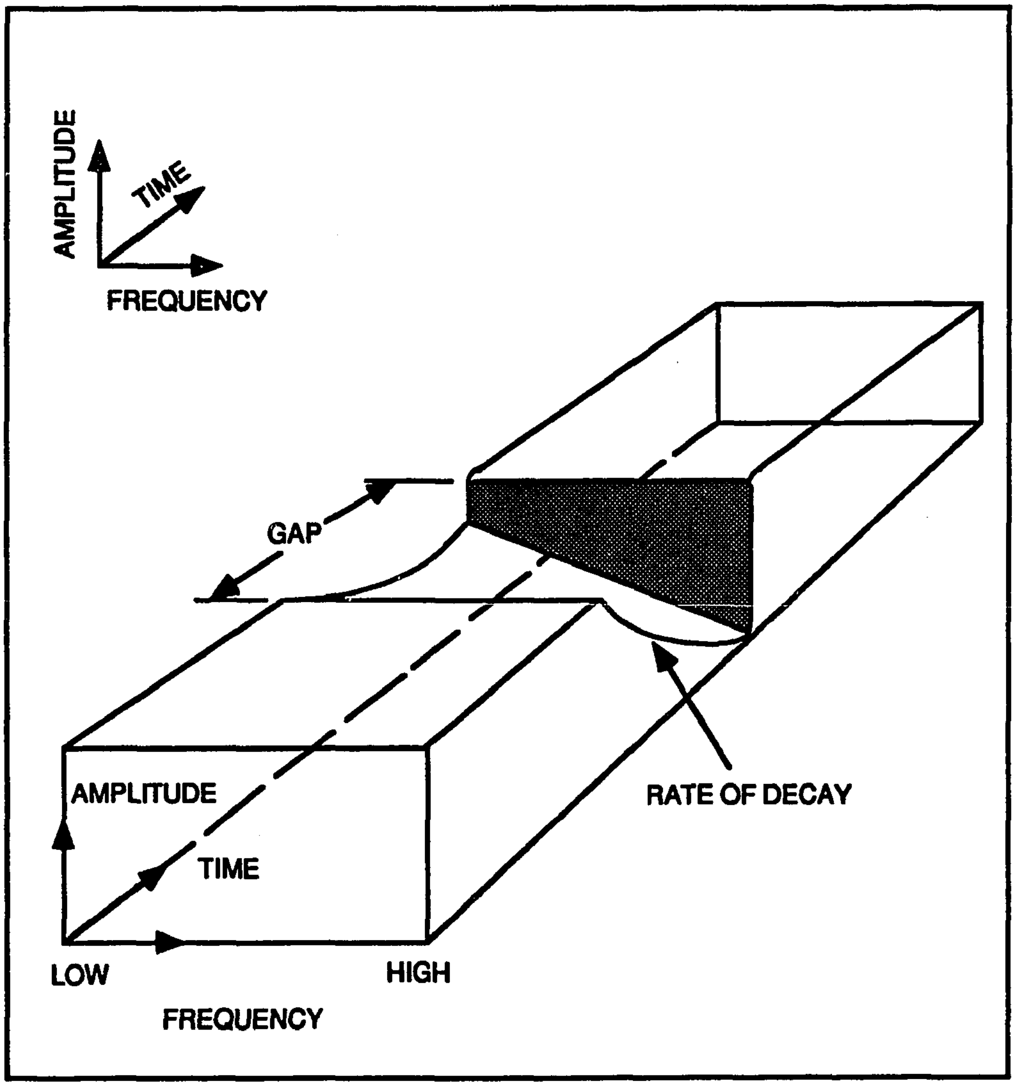

Figure 3.1. Schematic representation of the rate of decay of sensation for a brief gap in white noise. The low frequencies are assumed to decay at a slow rate and their decay is, therefore, represented by a shallow curve. The high frequencies are assumed to decay at a fast rate and their decay is, therefore, represented by the steep curve. It is assumed that the rate of decay increases with increasing frequency. Thus, the curves are steeper with increasing frequency. The filled area represents the increase of sensation to the level of the second marker when the latter is turned on.

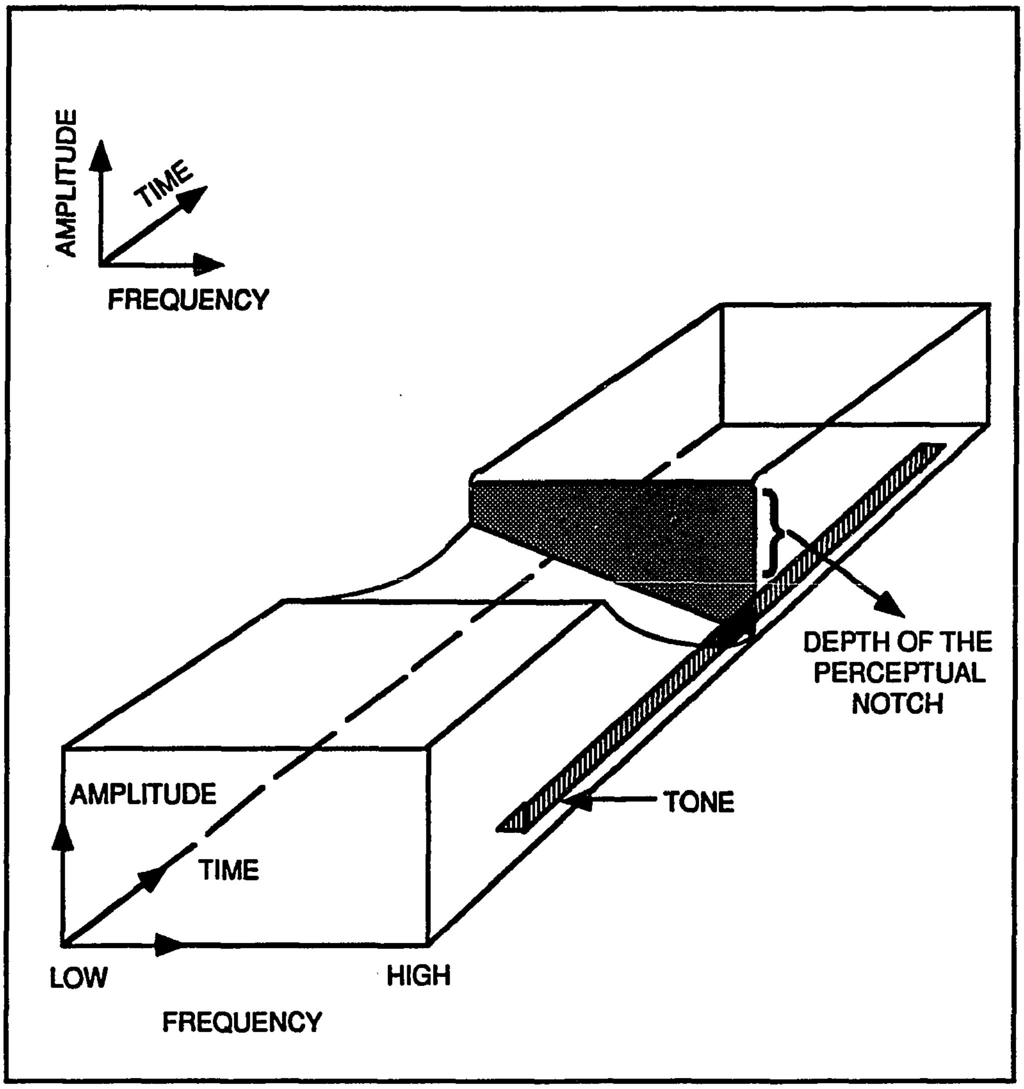

Figure 3.2. Schematic representation of a tone in a gap in white noise. The threshold of audibility for the tone is determined by its level in relation to the decaying sensation of the noise at its surrounding frequencies. The threshold of audibility of the tone in the gap in relation to its threshold in continuous noise (with no gap) is represented by the depth of the "perceptual notch". As frequency increases, the depth of the perceptual notch increases as well and the threshold of the tone decreases.

Methodology

Subjects

Five normally hearing listeners with no reported history of hearing disorders took part in this study. The subjects were between the ages of 23 and 40 years. Two were male and three were female. To determine that subjects' hearing status was within normal limits, air conduction thresholds were obtained at frequencies 250 through 8000 Hz prior to the first testing session. Subjects' thresholds did not exceed 15 dB HL in the tested ear. Individual subject information including the thresholds obtained at each frequency may be found in Appendix A.

Stimuli

The stimuli used in a single trial of this experiment are illustrated in Figure 3.3. Each trial consisted of a test stimulus and a control stimulus. The total duration of each stimulus was approximately 800 msec. These duration values are in keeping with the finding that gap thresholds are unaffected by the length of the surrounding noise bursts if their durations are greater than 20 msec (Penner, 1977).

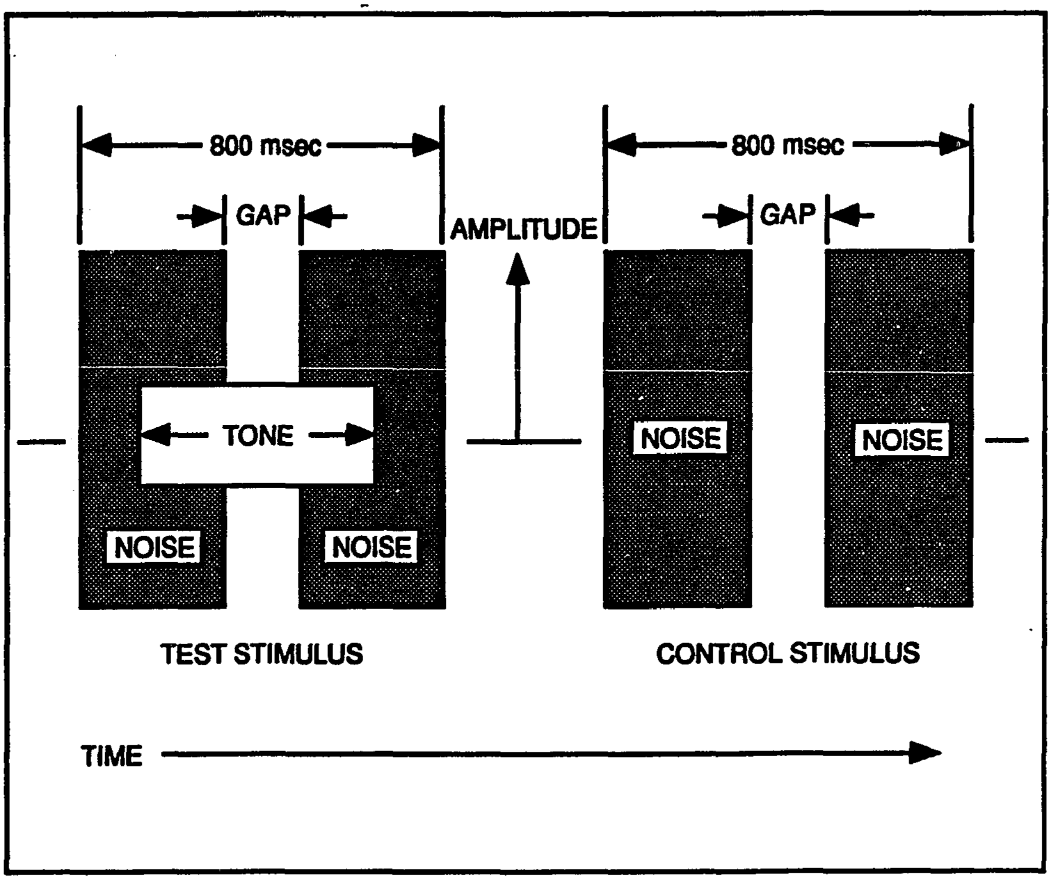

Figure 3.3. Schematic Representation of the Stimuli used in a Single Trial.

The test stimulus consisted of two white noise bursts separated by a gap. A sine-wave probe tone was switched on during the first noise burst, remained on throughout the gap, and switched off during the second noise burst. The onset and offset of the tone during the noise ensured that any transients produced by the switching would be masked by the noise. Thus, we were confident that the onset and offset of the tone would not serve as cues for the detection of the tone. Four probe tone frequencies were used: 0.5, 1.0, 2.0 and 4.0 KHz.

The control stimulus also consisted of two white noise bursts separated by a gap. The width of the gap was the same as that of the test stimulus. No probe tones where introduced in this stimulus. The interstimuli interval was set to 600 msec.

Nine nominal gap widths were used: 0, 6, 16, 26, 36, 46, 66, 86, and 106 msec. Calibration measurements showed that the actual gaps were approximately 7.5% longer. Thus, the actual gap values were: 0, 6.5, 17.20, 28.0, 39.00, 49.5, 71.0, 92.5 and 114.0 msec. Threshold was also measured in quiet (i.e., an infinitely long gap).

The intensity level of the noise was 70 dB SPL as measured by a Bruel & Kjaer (B & K) Sound Level Meter (Type 1613) at the output of the headphone using a B & K Artificial Ear (6 cc coupler, Type 4152) and a B & K Microphone (Type 4144). The level of the noise remained constant throughout all trials and testing sessions. Although this was not a gap detection task, it was important to use a noise level high enough so that gap threshold would not be dependent on intensity. Previous studies (Plomp, 1964; Boothroyd, 1973; Fitzgibbons, 1983; and Shailer and Moore, 1983) have shown that a minimum level of 30 to 40 dB SL (approximately 40 to 50 dB SPL) is necessary to attain this goal. Hence the choice of 70 dB SPL which is well above this level but not uncomfortably loud.

Instrumentation

The instrumentation used in this study is illustrated in Figure 3.4. Pure-tones were generated by a voltage control oscillator (Coulbourn, S24-05). The frequency of the pure tone was controlled by an Apple II computer via a Digital to Analog output of a Cyborg ISAAC interface. The voltage was then fed to the analog input of the voltage control oscillator (VCO). A frequency counter was used to ensure that all tested frequencies were within 1% of the target frequency.

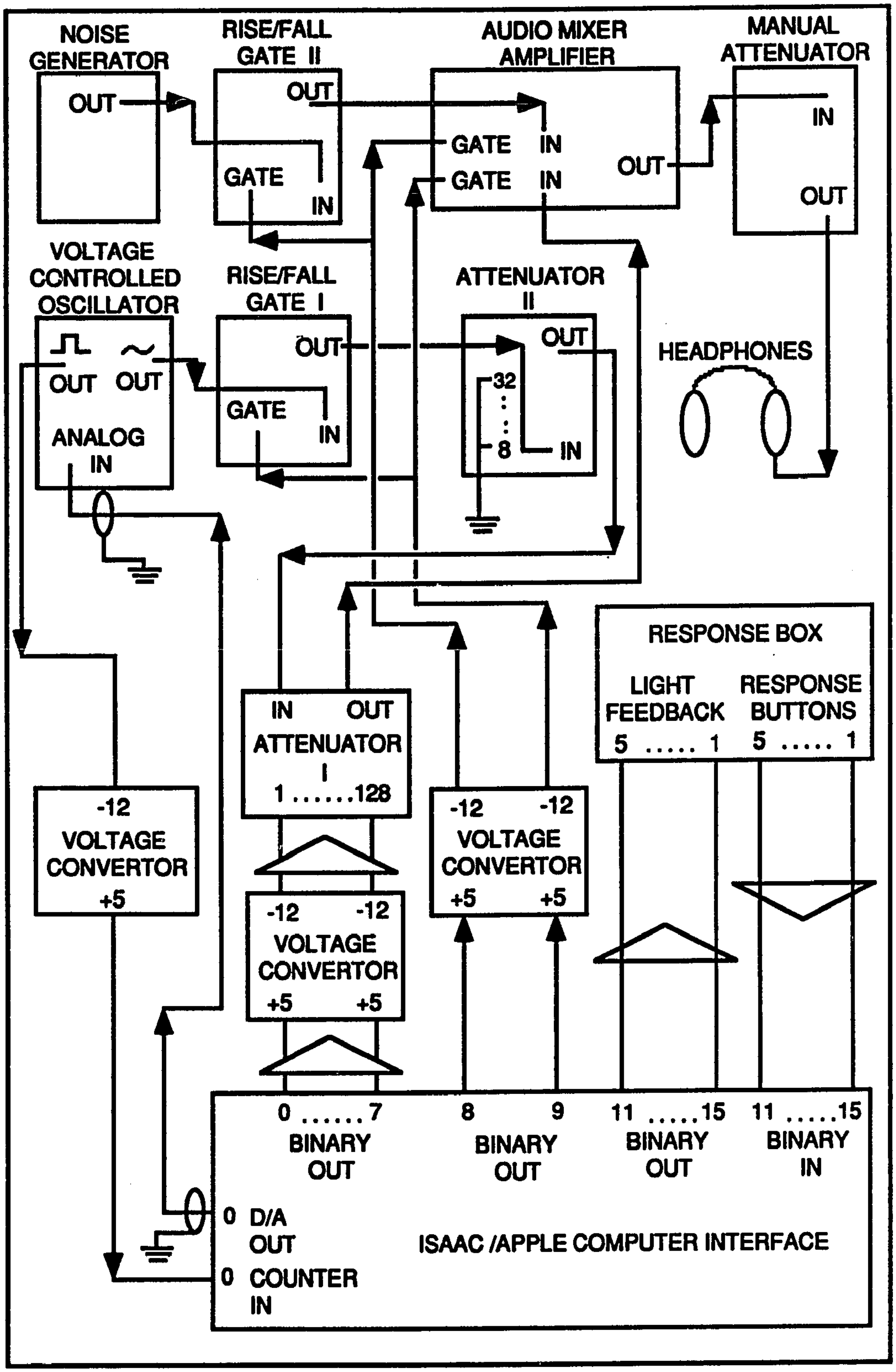

Figure 3.4. Block Diagram of the Instrumentation used in the Experiment.

The output sine wave signals were then gated by an electronic switch (Coulbourn, S84-04) with a rise/fall time of 10 msec, attenuated via a fixed attenuator (Coulbourn, S85-08), further attenuated by a programmable attenuator (Coulbourn, S85-08) , and fed into an audio/mixer amplifier (Coulbourn, S82-24). The electronic switch and the programmable attenuator were controlled by a digital output of the ISAAC interface.

White noise was generated by a noise generator (Coulbourn, S81-02). The noise was gated by another electronic switch (Coulbourn, S84-04) with a rise/fall time of 0.1 msec, and fed to the audio/mixer amplifier (S82-24). The mixed signal was fed into a manual attenuator (Coulbourn, S85-02). The manual attenuator was set to 50 dB. The signal was then fed to headphones (Telephonics, TDH-49 serial no. 10343), so as to produce 70 dB SPL at the earphones (as measured in a 6cc coupler).

Subjects' responses were entered via a response box to the digital input of the ISAAC interface. Feedback was administered via the digital outputs of the ISAAC interface to lights on the response box.

The gating and the attenuation of the signals were also controlled by the digital output of the ISAAC interface. To match the 5 volt logic of the interface to the 12 volt logic of the Coulbourn modules, logic convertors were used (Coulbourn, S62-06).

Calibration of Experimental Stimuli

Measurement and adjustment of the maximum levels of the experimental stimuli was performed before each test session. The programmable and manual attenuators were set to zero. The tone generator was then adjusted so that the r.m.s. output of the probe tone at the mixer/amplifier was 300 millivolts. Similarly, the noise generator was adjusted so that the r.m.s. output of the noise at the mixer/amplifier was 1 volt. The r.m.s. output levels of the tone and noise were obtained using an A.C. millivoltmeter (GSC, 1289) which was connected to the mixer/amplifier.

Initial calibration of the experimental stimuli included measurement of frequency, durations of control stimulus, test stimulus, and gap, and the linearity of the fixed, programmable and manual attenuators. The measured frequencies of a sample of probe tones were found to be within 1% of the expected frequencies. Further details will be found in Appendix B.

A consistent difference of less than 3% was found between the measured durations of the control and test stimuli in a sample of trials. Previous studies have reported that the normal auditory system can detect a minimum of a 10% change in duration of long stimuli (>500 msec) (e.g. Small and Campbell, 1962; Henry, 1948). It was, therefore, believed that subjects could not detect the measured 3% change in duration, and that the possibility that the test described here measured just noticeable changes in duration rather than the threshold of audibility of the probe tone was negated. Details of the duration measurements of control and test stimuli will be found in Appendix C.

The duration of the gaps was found to be consistently larger (average of 7.5 %), than the expected values. From here on, only the actual (measured) gap widths will be mentioned. Details of the gap duration measurements are found in Appendix D.

The fixed, programmable and manual attenuators performed as expected. Details on the linearity measurements of the attenuators are found in Appendix E. The frequency response of the TDH-49 earphone was measured and is shown in Figure 3.5.

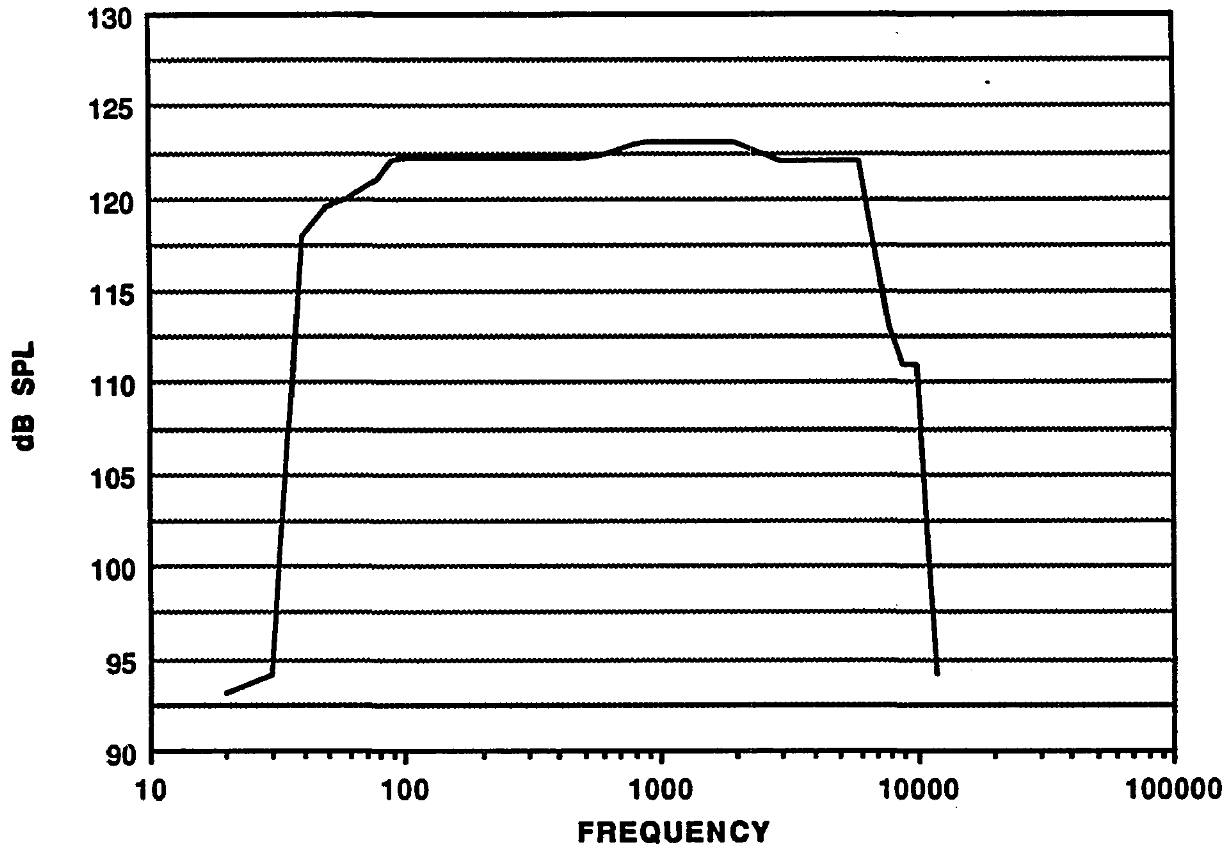

Figure 3.5. Frequency response of TDH 49 headphones used in the experiment.

Calibration measurements were repeated approximately once a month.

The noise floor level of the mixer/amplifier was measured to be 68 dB less than the highest level of the probe tone. When testing the probe tone's threshold in quiet, the manual attenuator following the mixer/amplifier was set to 80 dB thus lowering the noise floor below subjects' thresholds. Under all other test conditions, the manual attenuator was set to an attenuation level of 50 dB. In these conditions, the best tone threshold was never less than 10 dB above the noise floor and was usually much higher.

Procedure

Probe tone thresholds were obtained using a 2-interval, 2-alternative, forced choice (2I2AFC) adaptive procedure. Subjects were presented with a trial which consisted of a test stimulus and a control stimulus as shown in Figure 3.3. Subjects were required to determine which of the two noise bursts contained the pure tone. The forced-choice procedure (rather than a Yes/No task) was chosen because it reduces the effects of subjects' decision criteria on measured threshold. The binary procedure (rather than a 3 or more interval procedure) was chosen because it requires the subjects to identify the target signal (the probe tone in this case), and not simply to detect a difference.

The position of the probe tones (in stimulus 1 or 2) was determined randomly by the computer. No limitations were imposed on the number of consecutive times the probe tone was selected for a particular stimulus position. Random selection of the position of the probe tone was done to eliminate the effect of responding due to the expectation of a change in the test stimulus' position from trial to trial.

The adaptive procedure used to obtain the probe tone thresholds was a "2-down 1-up" adaptive procedure. That is, for every two consecutive correct responses the level of the tone decreased, and for every one incorrect response - the level of the probe tone was increased. This procedure estimates the 71% point on the psychometric function (Levitt, 1971), or the 42% point after correction for random guessing.

The probe tone level at the first presentation of a run was high enough for the listeners to be able to clearly determine in which of the two intervals of noise it was presented. After two consecutive correct responses the level of the tone was attenuated by a starting step size of 16 dB. If the listener responded correctly in two consecutive trials, the level of the tone was again attenuated by 16 dB. If, on the other hand, the listener responded incorrectly on one of two consecutive trials, the level of the tone was increased by half of the starting step size, i.e., by 8 dB. At every reversal, that is, a change in the direction of the probe tone's level (an increase after attenuation or vice versa), the step size was halved until a minimum step size of 2 dB was reached.

Testing was continued until 7 reversals at minimum step size were completed. The number of reversals was chosen as a compromise between test-retest reliability and testing time. The mean of the levels of attenuation at the 7 reversals was taken as the threshold estimate (in dB) for one run. An example of the computer's printout after a run is shown in Appendix F.

For each frequency and gap duration, excluding gap = 0 msec, 4 or more runs were completed until a standard error of the mean of all runs was 1 dB or less. For gap widths equal to 0 msec, thresholds were obtained until the standard error of the mean of all runs was 0.5 dB. It was important to obtain a more reliable measure at 0 msec gap because the 0 msec gap served as a reference for threshold improvement with the introduction of the gaps (see Chapter 4).

Subjects whose standard errors were less than the designated values did not have to replicate each measure more than 4 times. Those subjects, however, whose thresholds showed improvement with time, and/or great variability, were tested until the designated standard error values were reached. For each frequency and gap duration the mean of the runs was taken as the estimated probe tone threshold re dB attenuation. This procedure was controlled by software written in Labsoft (which is an extended version of Applesoft Basic) specifically for this study. See Appendix G for a complete listing of the program.

In each trial, one of two lights placed in front of the subject was lit to mark the onset of the first and second stimulus, respectively. The purpose of "marking" the onset of the stimuli with lights was to draw the listeners' attention to the auditorily presented stimuli without affecting their listening task.

Each subject was seated in a sound-attenuated chamber, given written instructions prior to testing (see Appendix H), and listened monaurally under TDH-49 headphones TDH-49. Stimuli were delivered to the right ear of three subjects and to the left ear of the remaining two subjects.

Subjects keyed in their responses via a response box placed in front of them. Feedback was administered by means of lights after each trial. That is, after the subject responded, the light corresponding to the correct response (i.e., light #1 or light #2) was illuminated on the response box.

Every subject received an hour of training before the actual data were collected. In a given testing session, each subject was tested at all frequencies with one gap duration. This was done to ensure that variables affecting the subject's results at the time of testing would not be confounded with the effects of frequency. The gap value to be tested at each session was chosen randomly. Within a test session the order of frequency presentation was also random. Each testing session lasted approximately 20-30 minutes. To avoid fatigue effects, subjects were not tested for more than two consecutive sessions (i.e. 40 minutes).

CHAPTER 4

RESULTS

Appendix I contains the pure-tone thresholds, in dB attenuation (re the maximum stimulus level available from the test equipment), obtained for each frequency and gap. Data are shown for each subject at each run. Also shown are the means, standard deviations, and standard errors for each subject, collapsed across runs.

The data will be examined in two ways: (1) threshold of audibility of the tone in relation to its value in noise (i.e., perceptual notch depth) as a function of gap size and frequency, and (2) gap size as a function of notch depth and frequency.

Analysis of the data using notch depth as the dependent variable

One measure of the subject's temporal acuity is the mean pure-tone threshold at a given gap width and frequency in relation to the threshold value in continuous noise. Thus, for a given gap width, the difference between thresholds in the gap and no gap conditions, reflects the depth of the perceptual notch.

Greater depth suggests better temporal acuity, i.e., faster decay of sensation. Mean threshold of audibility for each frequency relative to the threshold in noise are shown in Table 4.1 for each subject. Also shown are the group means and standard deviations. These data are illustrated in Figure 4.1. Table 4.1 and Figure 4.1 also show pure-tone thresholds in quiet.

Table 4.2 shows the results of both a repeated-measures and a simple 3-way analysis of variance in the data. (Note that the simple 3-way analysis of variance was included because of interest in inter-subject differences which will be discussed at the end of this chapter). The most important findings of this analysis are that the effect of frequency is highly significant (F (3,12) = 30.69, $p<0.01$), and that the interaction between frequency and gap width is highly significant (F (21,84) = 3.62, $p<0.01$). The highly significant effect of gap (F (7,28) = 249.71, $p<<0.01$) is an expected finding because the depth of the perceptual "notch" was obliged to increase with increasing gap width.

Table 4.1

Mean Threshold of Tones (in dB attenuation) re: Threshold of Tone in Continuous Noise for each Gap, Frequency and Subject. Also shown are Group Means and Standard Deviations.

		ACTUAL GAP (in msec)								
FREQ (KHz)	SUBJ	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
0.5	LK	-2.66	0.03	2.16	10.59	12.69	22.22	22.92	24.34	32.74
0.5	EL	1.48	0.85	7.26	7.80	16.13	17.88	19.90	19.30	25.96
0.5	MB	2.03	4.48	7.76	9.88	14.20	23.63	26.18	28.35	38.85
0.5	OE	2.20	6.95	8.80	12.20	13.28	16.00	19.10	23.25	29.45
0.5	LH	1.26	2.60	3.62	5.55	7.80	13.52	14.18	23.32	33.84
1.0	LK	0.75	9.40	17.14	23.49	28.32	32.47	35.90	36.50	45.51
1.0	EL	-2.08	8.90	16.32	19.43	21.05	25.40	27.19	32.35	39.53
1.0	MB	-0.05	8.12	15.67	18.80	22.62	28.20	33.70	35.35	47.65
1.0	OE	0.31	14.21	23.14	23.94	27.24	32.41	34.56	40.04	48.24
1.0	LH	4.35	7.13	12.53	14.91	17.38	23.18	27.87	27.45	38.03
2.0	LK	0.51	15.08	21.91	25.01	27.26	32.36	31.83	35.74	40.48
2.0	EL	2.90	10.57	14.94	19.96	20.40	25.72	28.87	32.12	39.90
2.0	MB	0.97	13.80	17.37	23.12	24.22	28.45	33.62	34.62	46.67
2.0	OE	5.06	19.55	21.78	23.88	28.56	28.53	30.58	31.61	44.58
2.0	LH	-1.64	7.76	14.49	15.94	20.95	26.41	33.56	35.09	51.26
4.0	LK	5.33	20.71	21.50	24.59	27.54	30.11	36.31	36.04	43.08
4.0	EL	1.45	18.35	22.90	24.17	28.43	32.17	36.52	42.82	48.50
4.0	MB	1.63	11.15	14.90	18.63	25.55	27.34	32.20	31.55	42.70
4.0	OE	7.74	24.07	26.47	26.65	35.83	38.57	40.77	40.37	55.35
4.0	LH	4.36	11.51	16.86	17.86	18.74	29.24	24.90	25.91	38.91
GRAND MEAN										
0.5		0.86	2.98	5.92	9.20	12.82	18.65	20.46	23.71	32.17
(SD)		(2.01)	(2.80)	(2.87)	(2.58)	(3.09)	(4.23)	(4.48)	(3.23)	(4.84)
1.0		0.66	9.55	16.96	20.11	23.32	28.33	31.84	34.34	43.79
(SD)		(2.33)	(2.74)	(3.87)	(3.72)	(4.51)	(4.15)	(4.02)	(4.73)	(4.72)
2.0		1.56	13.35	18.10	21.58	24.28	28.29	31.69	33.84	44.58
(SD)		(2.54)	(4.49)	(3.59)	(3.67)	(3.65)	(2.59)	(2.03)	(1.85)	(4.68)
4.0		4.10	17.16	20.53	22.38	27.22	31.49	34.74	35.34	45.71
(SD)		(2.64)	(5.70)	(4.67)	(3.90)	(6.13)	(4.32)	(5.99)	(6.81)	(6.38)

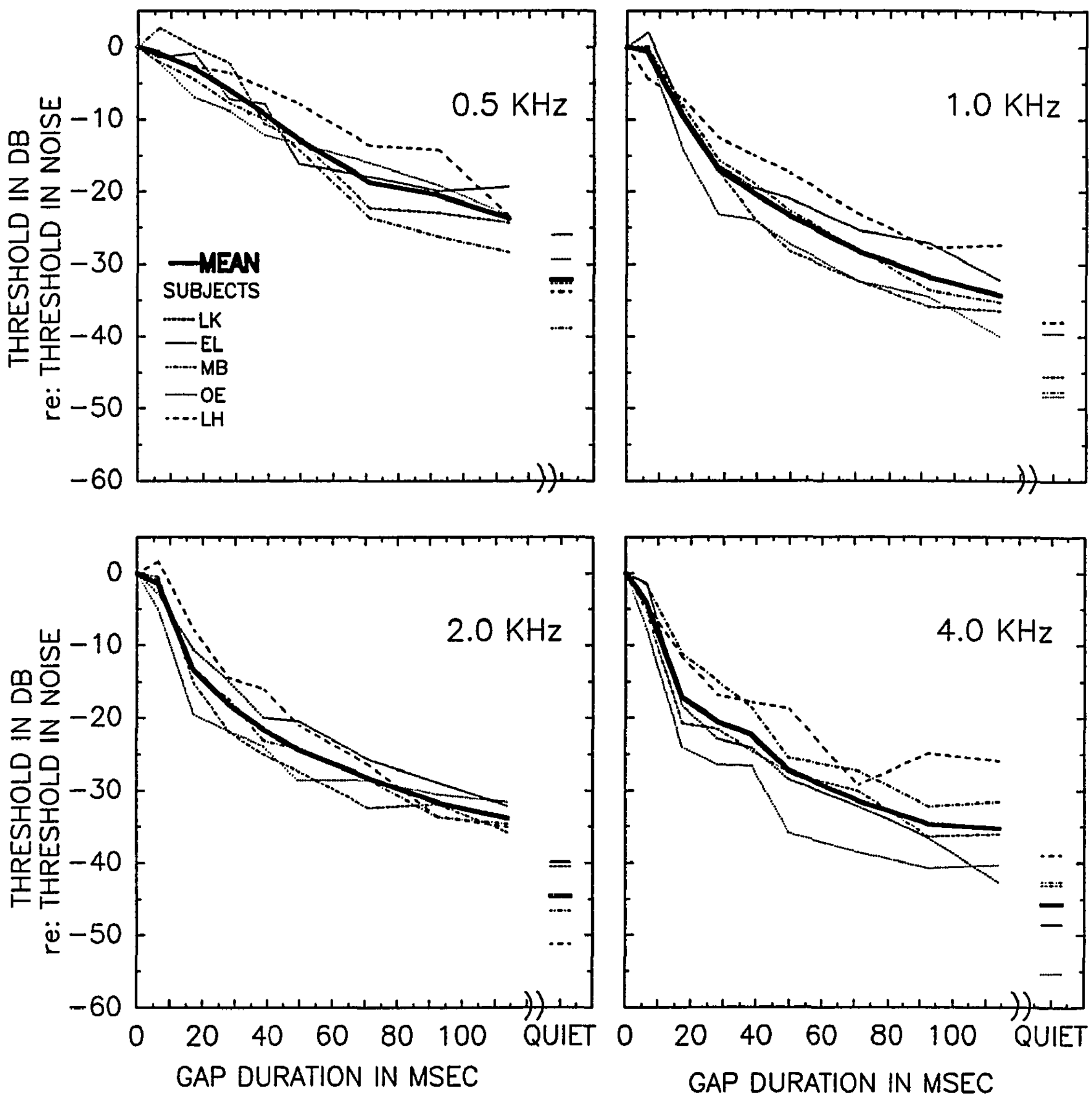

Figure 4.1 Thresholds re: Thresholds in Noise for Each Frequency and Subject. The Thick Solid Lines Represent the Mean Threshold of 5 Subjects at Each Frequency.

Table 4.2

ANOVA Summary Table for the Effects of Frequency, Gap and Subject for Gaps 6.5 msec through 114.0 msec.

SOURCE	df	MEAN SQUARE	F Repeated Measures	F Simple 3-Way
FREQUENCY (F)	3	1135.81	30.69**	195.49**
GAP (G)	7	2050.94	249.71**	353.00**
SUBJECT (S)	4	207.42	---	35.68**
F x G	21	21.05	3.62**	3.62**
F x S	12	37.01	---	6.39**
G x S	28	8.21	---	1.41
F x G x S	84	5.81		

** Significant at $p < 0.01$

To illustrate the interaction between frequency and gap width, the means for each gap width at each frequency (collapsed over subjects) are shown in Figure 4.2. It will be seen that the effect, on threshold, of changing gap width is different for different frequencies. Specifically, for 0.5 KHz the threshold falls at a fairly constant rate as gap width increases. For 1, 2, and 4 KHz, however, the threshold falls rapidly as gap width increases from 0 to 28.0 msec, but more gradually thereafter. After 28.0 msec the 4 curves are essentially parallel.

The observation that the functions for the 4 frequencies become parallel above 28.0 msec suggests that the frequency by gap interaction found in the ANOVA is due only to what happens at the smallest gap widths. To test this, two additional analyses were performed. One was an analysis of the variance in the data for gap widths 6.5 and 17.2 msec and the second was an analysis of the variance in the data for gap widths 28.0 through 114.0 msec.

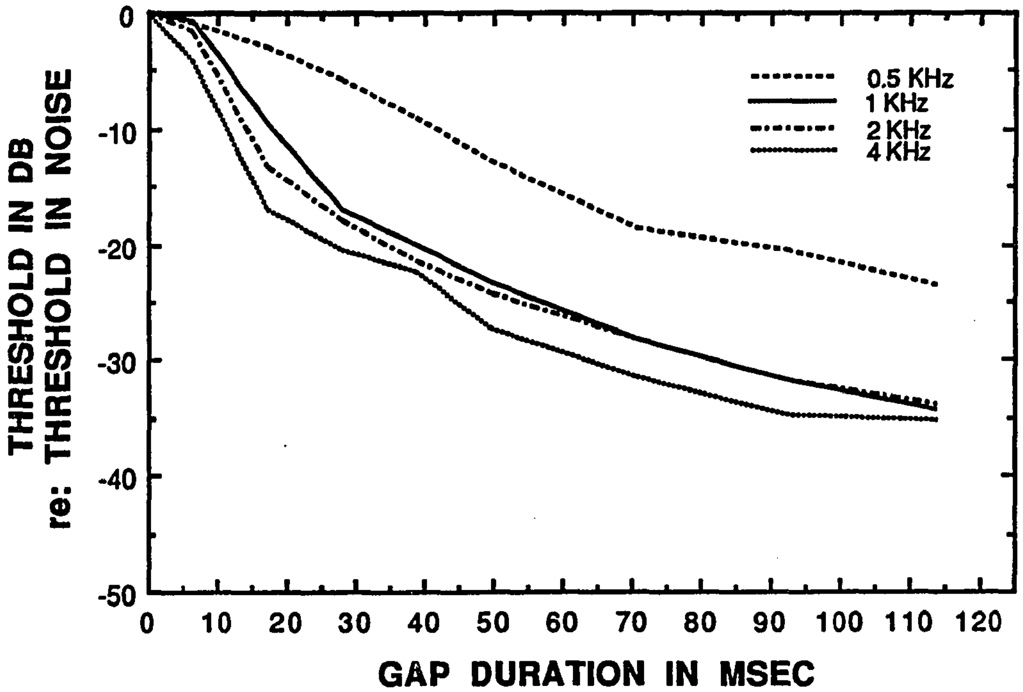

Figure 4.2. Threshold in dB re: Threshold in Noise for Each Frequency as a Function of Gap Width Averaged across Subjects.

The results of the first analysis in the data for gap widths 6.5 and 17.20 msec are shown in Table 4.3. It can be seen that the interaction between frequency and gap width is, as expected, significant ($\underline{F}$ (3,12) = 16.59, $\underline{p}<0.01$). In other words, for the high frequencies, there is a significantly faster increase in the depth of the "notch" as gap width increases from 6.5 msec to 17.2 msec. For the low frequencies, however, the depth of the "notch" does not seem to vary significantly with increase of gap width to 17.2 msec.

The results of the second analysis in the data for gap widths 28.0 and 114.0 msec are shown in Table 4.4. It can be seen that for gap widths of 28.0 msec and greater, the frequency by gap interaction is, indeed, not significant ($\underline{F}$ (15,60) = 0.44, $\underline{p}>0.05$). That is, as the gap widens (above 17.2 msec), the rate of change in the depth of the "notch" (as measured in decibels) is constant across frequencies.

Table 4.3

ANOVA Summary Table for the Effects of Frequency, Gap and Subject for Gaps 6.5 and 17.2 Msec.

SOURCE	df	MEAN SQUARE	F Repeated Measures	F Simple 3-Way
FREQUENCY (F)	3	135.60	14.12**	37.67**
GAP (G)	1	803.89	56.58**	223.30**
SUBJECT (S)	4	37.06	---	10.31**
F x G	3	59.66	16.59**	16.59**
F x S	12	9.60	---	2.67(*)
G x S	4	14.21	---	3.95*
F x G x S	12	3.60		

(*) Approaching the 0.05 significance level

* Significant at $p < 0.05$

** Significant at $p < 0.01$

Table 4.4

ANOVA Summary Table for the Effects of Frequency, Gap and Subject for Gaps 28.0 to 114.0 Msec.

SOURCE	df	MEAN SQUARE	F Repeated Measures	F Simple 3-Way
FREQUENCY (F)	3	1076.76	27.32**	214.49**
GAP (G)	5	839.08	175.39**	167.15**
SUBJECT (S)	4	189.71	---	37.82**
F x G	15	2.23	0.44	0.44
F x S	12	39.42	---	7.86**
G x S	20	4.78	---	0.95
F x G x S	60	5.02		

** Significant at $p < 0.01$

Because the interaction between frequency and gap width is significant only for the smaller gap widths, the effect of frequency was explored separately for gap widths 6.5 and 17.2 msec and for gap widths 28.0 through 114.0 msec. In the repeated-measures analysis of variance in the data for gap widths 6.5 and 17.2 msec, (Table 4.3) the effect of frequency was found to be significant (F (3,12) = 14.12, $p<0.01$). The Student Newman-Keuls procedure was applied to determine the significance of differences among the means for the individual frequencies. The results are summarized in Table 4.5. It will be seen that not only is the mean of 0.5 KHz significantly lower than the means of 1.0, 2.0, and 4.0 KHz, but also, the mean of 4.0 KHz is significantly higher than the means of 0.5, 1.0 and 2.0 KHz. These results suggest that for brief gaps (<28.0 msec), there is a frequency effect from 0.5 to 1.0 KHz, and some evidence of a frequency effect above 1.0 KHz.

Table 4.5

Results of Student-Newman Keuls Procedure Applied to Differences Between the Frequency Means for Gaps 6.5 and 17.2 Msec.

	FREQUENCY (KHz)			
	0.5	1.0	2.0	4.0
FREQUENCY (KHz)				
0.5	-	3.18*	5.53**	8.71**
1.0	-	-	2.35	5.53**
2.0	-	-	-	3.17*

* Significant at $p < 0.05$

** Significant at $p < 0.01$

Critical Differences:

Rank	$p < 0.05$	$p < 0.01$
2	3.02	4.23
3	3.67	4.94
4	4.12	5.39

In the repeated-measures analysis of variance in the data for gap widths 28.0 through 114.0 msec, the effect of frequency is also significant ($\underline{F}$ (3,12) = 27.32, $\underline{p}$<0.01, Table 4.4). Results of the Student Newman-Keuls procedure applied to determine the significance of differences among the means for the individual frequencies are summarized in Table 4.6. As found previously, the perceptual "notch" is significantly less at 0.5 KHz than at 1.0, 2.0, and 4.0 KHz. For 1.0, 2.0, and 4.0 KHz, however, the depths of the perceptual "notches" were not found to be significantly different. These results suggest that for gap widths equal to or greater than 28.0 msec, temporal acuity, as measured here, improves from 0.5 to 1.0 KHz but remains constant for frequencies 1.0, 2.0, and 4.0 KHz.

In the simple 3-way analysis of variance in the data for gaps 6.5 to 114.0 msec (Table 4.2), there was also evidence of a significant frequency by subject interaction ($\underline{F}$ (12,84) = 6.39, $\underline{p}$<0.01). Because the interaction between frequency and gap width is significant only for brief gaps, frequency by subject interaction was explored separately for gap widths 6.5 to 17.2 msec and 28.0 to 114.0 msec.

Table 4.6

Results of Student-Newman Keuls Procedure Applied to Differences Between the Frequency Means for Gaps 28.0 to 114.0 Msec.

	FREQUENCY (KHz)			
	0.5	1.0	2.0	4.0
FREQUENCY (KHz)				
0.5	-	10.69**	11.71**	13.39**
1.0	-	-	0.48	2.70
2.0	-	-	-	2.22

** Significant at $p < 0.01$

Critical Differences:

Rank	$p < 0.05$	$p < 0.01$
2	3.53	4.95
3	4.32	5.78
4	4.81	6.30

For gaps 6.5 to 17.2 msec, the interaction between frequency and subject approaches (but does not reach) the 0.05 significance level (F (12,12) = 2.67, p>0.05, Table 4.3). Thus, for brief gaps, subjects preformed similarly across frequencies.

For gaps 28.0 through 114.0 msec, however, the interaction between frequency and subject is highly significant (F (12,60) = 7.86, p<0.01, Table 4.4). This is further illustrated in Figure 4.3. Examination of this figure reveals that for all subjects the average depth of the "notch" was greater at 1.0 KHz than at 0.5 KHz. Also, "notch" depth did not increase at 2.0 KHz but remained similar to the one observed at 1.0 KHz. At 4.0 KHz, however, the variability of the data increased. For three subjects (LH, LK & MB), the increase of frequency to 4.0 KHz did not affect the depth of the "notch". For the remaining two subjects (OE & EL), however, the increase of frequency to 4.0 KHz resulted in a relatively large increase in "notch" depth. These results suggest that all subjects showed a similar pattern of change in the depth of the perceptual "notch" as frequency increased from 0.5 to 2.0 KHz. At 4.0 KHz, however, subject variability increased. Two subjects showed an increase in the depth

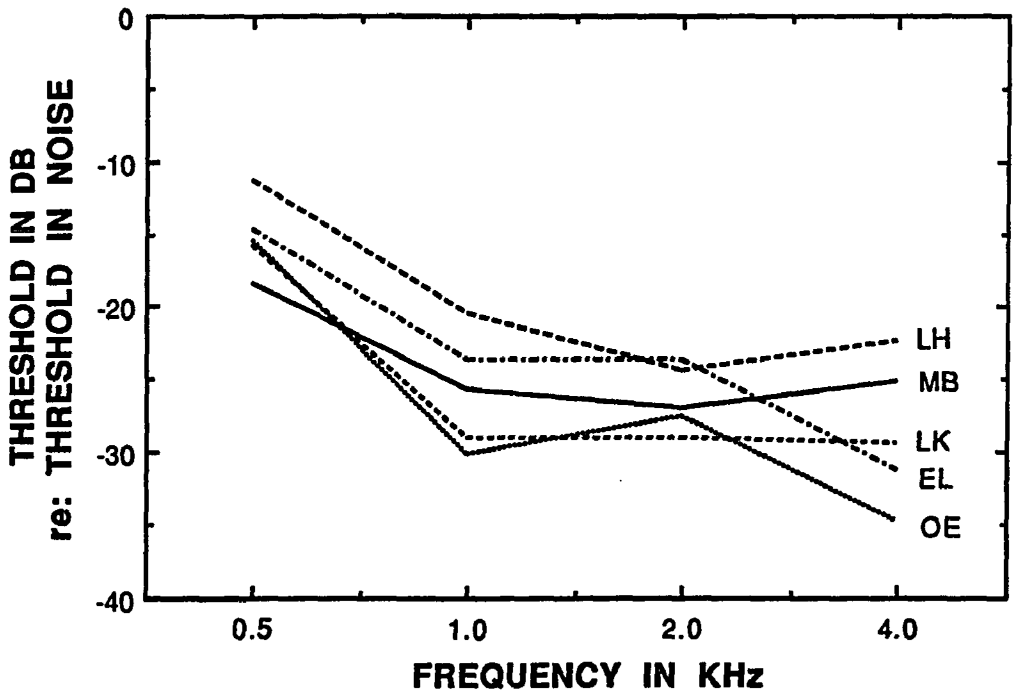

Figure 4.3. Threshold in dB re: Threshold in Noise for Each Subject as a Function of Frequency. Data are Averaged Across Gaps 27.95 to 113.95 Msec.

of the "notch" whereas the remaining subjects showed an unchanged "notch" depth. Note that this was a simple 3-way analysis where subjects were treated as a fixed effect. Thus, in computing the F-ratio for the interaction between frequency and subject, unknown sources of error variance appeared in the numerator and in the denominator and the estimate of the F-ratio is negatively biased. A significantly high F-ratio, therefore, may be taken as strong evidence that a frequency by subject interaction exists.

Summary of results

1. When temporal acuity was measured in terms of the depth of the perceptual notch resulting from a brief gap in a noise stimulus, there was evidence of frequency dependence.

2. In particular, temporal acuity improved considerably from 0.5 to 1.0 KHz.

3. There was also evidence of weak frequency dependence between 1.0 and 4.0 KHz for very brief gaps (<28.0 msec) and for some subjects.

Preliminary Discussion of Results

In this section, thresholds of pure-tones at different gap widths were calculated in relation to their thresholds in continuous noise (i.e., no gap) as the measure of temporal acuity. It is assumed that this measure reflected the depth of the perceptual "notch" resulting from the decay of sensation during the physical gap in the noise. A lower threshold for a fixed gap width suggests a deeper "notch" in sensation and better temporal acuity, whereas a higher threshold suggests a shallower "notch" and poorer temporal acuity. The results as analyzed here suggest two separate processes of temporal acuity. One process occurs at very small gap widths and shows a clear frequency dependence between 0.5 and 4.0 KHz as hypothesized by the simple auditory-filter model. In this process there seems to be a sharpening effect at the higher frequencies which cause the depth of the "notch" to increase rapidly at first and thus improve temporal acuity. The second process occurs at higher gap widths and shows frequency dependence only between 0.5 and 1.0 KHz. The "notch" depths at 1.0, 2.0, and 4.0 KHz are very similar suggesting no frequency dependence above 1 KHz. This last finding, however, was subject-dependent. Two of the five subjects showed an improvement of temporal acuity between 2.0 and 4.0 KHz.

It could be argued that the depth of the perceptual "notch" is not the most appropriate measure of temporal acuity. In the following section, the experimental questions are re-addressed using an alternative measure.

Alternative analysis using gap width as the dependent variable

In the previous section, temporal acuity was assessed in terms of the depth of the perceptual notch for different gap sizes in noise stimuli. This was accomplished by calculating thresholds of pure-tones at different gap widths in relation to their thresholds in continuous noise (i.e., no gap). Using these data, frequency dependence of temporal acuity was observed at small gap widths (<28.0 msec) for frequencies 0.5 through 4.0 KHz. At greater gap widths, there was strong evidence of frequency dependence between 0.5 and 1.0 KHz, but only weak, subject-specific evidence of frequency dependence above 1.0 KHz.

Note, that the "notch depth" measure is not a temporal measure but a decibel measure. In existing literature, temporal acuity is most commonly measured as the width of a just detectable gap, which is, of course, a temporal measure. In order to express the results of the present study in temporal units, the minimum gap required to detect a tone at a given "notch" level was assessed.

Gap widths required to achieve "notch" depths of 5, 10, 15, 20, 25 and 30 dB attenuation were calculated (interpolated) from the data shown in Table 4.1. The calculated values for each subject, frequency and "notch" depth and the means are shown in Table 4.7. It will be seen that for 0.5 KHz, a gap width of 114.0 msec was insufficient to achieve "notch" depths of 25 and 30 dB attenuation. Similarly, a 114.0 msec gap width was insufficient to achieve a "notch" depth of 30 dB attenuation at 1.0 and 4.0 KHz for one subject, LH. For these reasons, the remaining analyses and illustrations will relate to data for "notch" depths of 5, 10, 15, and 25 dB attenuation only.

Table 4.7

Gap Width in Milliseconds Required to Detect Tones in Notch Depths 5 to 30 dB.

FREQ (KHz)	SUBJ	NOTCH DEPTH (in dB Attenuation) 5	10	15	20	25	30
0.5	LK	31.32	37.25	54.30	64.79	-	-
0.5	EL	23.67	41.34	47.34	117.00	-	-
0.5	MB	18.79	38.98	51.15	61.75	81.70	-
0.5	OE	12.34	31.48	62.10	96.79	-	-
0.5	LH	35.10	57.14	94.27	105.22	-	-
1.0	LK	11.30	17.98	24.44	32.45	41.83	57.55
1.0	EL	12.90	18.68	25.42	42.22	67.61	103.34
1.0	MB	12.63	19.69	26.31	41.84	57.98	77.50
1.0	OE	9.82	13.42	18.08	23.68	41.91	60.13
1.0	LH	8.79	22.51	39.06	58.48	78.71	-
2.0	LK	9.53	12.96	16.40	24.40	38.02	60.20
2.0	EL	9.19	15.71	28.07	39.61	66.74	99.40
2.0	MB	9.59	13.49	20.36	32.52	53.14	76.95
2.0	OE	6.41	9.86	13.31	19.22	41.09	85.29
2.0	LH	13.51	20.53	31.47	46.80	64.29	80.99
4.0	LK	6.24	9.49	12.74	15.99	40.09	68.59
4.0	EL	8.55	11.51	14.47	20.83	40.65	57.85
4.0	MB	9.99	15.24	28.22	40.68	47.91	81.90
4.0	OE	4.77	7.83	10.90	13.96	21.16	42.35
4.0	LH	7.35	14.34	23.72	51.85	67.48	-
GRAND MEAN							
0.5		24.24	41.24	61.83	89.11	-	-
(SD)		(9.2)	(9.6)	(18.9)	(24.7)		
1.0		11.09	18.46	26.66	39.73	57.61	-
(SD)		(1.8)	(3.3)	(7.7)	(13.0)		
2.0		9.65	14.51	21.92	32.51	52.66	80.57
(SD)		(2.5)	(4.0)	(7.7)	(11.2)		
4.0		7.38	11.68	18.01	28.66	43.46	-
(SD)		(2.0)	(3.1)	(5.5)	(16.7)		

Figure 4.4 shows the gap width required to achieve a fixed "notch" depth for frequencies 0.5 through 4.0 KHz. In general, for every "notch" depth, a smaller gap width is required for 0.5 KHz than for 1.0 KHz. The parallel lines above 1.0 KHz, however, suggest that similar gap widths are required to maintain a fixed "notch" depth for frequencies 1.0, 2.0, and 4.0 KHz. The following statistical analyses provide further insight to these observations.

Table 4.8 shows the results of both a repeated-measures and a simple 3-way analysis of the variance for the data shown in Table 4.7. The effects of frequency and notch depth are found to be highly significant ($\underline{F}$ (3,12) = 57.52, $\underline{p}$<0.01, and $\underline{F}$ (3,12) = 39.13, $\underline{p}$<0.01, respectively). The interaction between frequency and notch depth ($\underline{F}$ (9,36) = 6.31, $\underline{p}$<0.01) was also found to be highly significant.

That there is an interaction between frequency and "notch" depth only for frequencies 0.5 and 1.0 KHz, but not for frequencies 1.0, 2.0, and 4.0 KHz, suggests that the frequency by "notch" depth interaction found in the ANOVA is due to the lower frequencies. Specifically, greater gap widths are required as the depth of the "notch" increases at 0.5 KHz than at frequencies above 0.5 KHz.

To test this last observation, two additional analyses were performed. The first was an analysis of the variance in the data for 0.5 and 1.0 KHz and its results are shown in Table 4.9. The second was an analysis of the variance in the data for frequencies 1.0, 2.0 and 4.0 KHz and its results are shown in Table 4.10.

It will be seen that a frequency by notch depth interaction was highly significant for 0.5 and 1.0 KHz ($F(3,12) = 5.97$, $p<0.01$), but not for frequencies 1.0, 2.0 and 4.0 KHz ($F(6,24) = 1.28$, $p>0.05$). These results suggest that as frequency increases from 0.5 to 1.0 KHz, the rate of change of gap duration is dependent on the depth of the "notch". For frequencies above 1.0 KHz, however, this is not the case. That is, the rate of change of gap duration with increasing frequency is independent of "notch" depth.

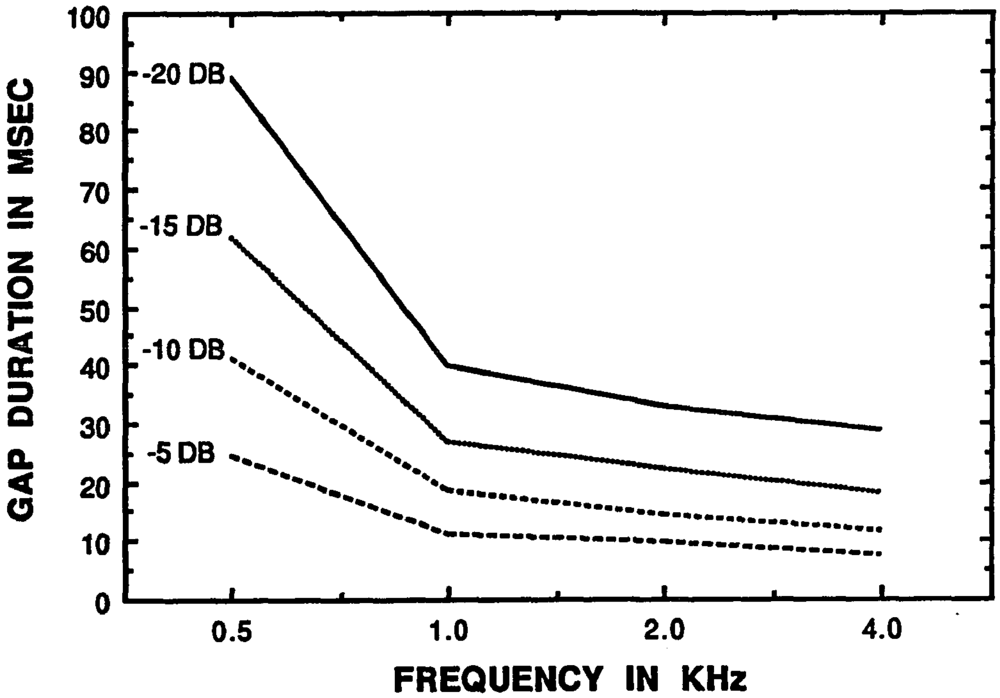

Figure 4.4. Gap Width (in Msec) Required to Detect Tones in Notch Depths of 5 to 20 dB as a Function of Frequency. Data are Mean of 5 Subjects.

Table 4.8

ANOVA Summary Table for the Effects of Frequency, Subject and Notch Depth. Data are Gap Widths Expressed in Msec.

SOURCE	df	MEAN SQUARE	F Repeated Measures	F Simple 3-Way
FREQUENCY (F)	3	6000.14	57.52**	96.81**
SUBJECT (S)	4	732.97	---	13.90**
NOTCH DEPTH (N)	3	4406.90	39.13**	71.10**
F x S	12	104.31	---	1.68
F x N	9	391.03	6.31**	6.31**
S x N	12	112.62	---	1.82(*)
F x S x N	36	61.98		

** Significant at $p < 0.01$

(*) Significant at $p < 0.1$

Table 4.9

ANOVA Summary Table for the Effects of Frequency, Subject and Notch Depth for Frequencies 0.5 and 1 KHz. Data are Expressed in Msec.

SOURCE	df	MEAN SQUARE	F Repeated Measures	F Simple 3-Way
FREQUENCY (F)	1	9072.75	57.18**	88.15**
SUBJECT (S)	4	535.92	---	5.21*
NOTCH DEPTH (N)	3	4042.40	31.75**	39.27**
F x S	4	158.67	---	1.54
F x N	3	614.93	5.97**	5.97**
N x S	12	127.32	---	1.24
F x S x N	12	102.93		

*Significant at $p < 0.05$

**Significant at $p < 0.01$

Table 4.10

ANOVA Summary Table for the Effects of Frequency, Notch Depth and Subject for Frequencies 1, 2, and 4 KHz. Data are Expressed in Msec.

SOURCE	df	MEAN SQUARE	F Repeated Measures	F Simple 3-Way
FREQUENCY (F)	2	287.23	10.23**	27.70**
SUBJECT (S)	4	430.57	---	41.52**
NOTCH DEPTH (N)	3	1649.53	19.95**	159.05**
F x S	8	28.07	---	2.71*
F x N	6	13.32	1.28	1.28
N x S	12	82.68	---	7.97**
F x S x N	24	10.37		

*Significant at $p < 0.05$

**Significant at $p < 0.01$

Because the interaction between frequency and "notch" depth is significant only for the lower frequencies, the effect of frequency was explored separately for frequencies 0.5 and 1.0 KHz and for frequencies 1.0, 2.0, and 4.0 KHz. A significant effect of frequency was found in both repeated measures analyses of variance ($F(1,4) = 57.18$, $p<0.01$, and $F(2,8) = (10.23)$, $p<0.01$, for frequencies 0.5 - 1.0 KHz and frequencies 1.0 - 4.0 KHz, respectively). In order to determine the significance of differences among the means for the individual frequencies (1.0 to 4.0 KHz), the Student Newman-Keuls procedure was applied. The results are summarized in Table 4.11. Overall, gap widths (in msec) for 1 KHz are significantly greater than those obtained at 2.0 and 4.0 KHz. The difference between 2.0 and 4.0 KHz failed to reach the 0.05 level of significance.

Gap widths for frequencies 1.0, 2.0 and 4.0 KHz were significantly different from those derived at 0.5 KHz. These results suggest frequency dependence of temporal resolution. Specifically, temporal acuity was found to be poorest at 0.5 KHz and significantly poorer at 1.0 KHz. It is not clear, however, that temporal acuity continues to improve between 2.0 and 4.0 KHz.

Table 4.11

Results of Student-Newman Keuls Procedure Applied to Differences Between the Means for Frequencies 1, 2, and 4 KHz. Data are Expressed in Msec.

FREQUENCY (KHz)	1.0	2.0	4.0
1.0	-	4.34*	7.55**
2.0	-	-	3.21

* Significant at $p < 0.05$

** Significant at $p < 0.01$

Critical Differences:

Rank	$p < 0.05$	$p < 0.01$
2	3.86	5.62
3	4.79	6.67

In the simple 3-way analysis of variance (Table 4.8), the interaction between frequency and subject failed to reach the 0.05 significance level ($F(3,12) = 1.98$, $p>0.05$). This is contrary to the finding of a strong frequency by subject interaction when temporal acuity was measured in decibels.

In observing the data in Table 4.7 it appears that the assumption of homogeneity of variance has been violated. To correct for this violation, a logarithmic transformation was applied to the data. Figure 4.5 shows the duration of the gap (in msec) expressed on a logarithmic scale as a function of frequency for each level of the perceptual notch averaged across the five subjects. It can be seen by the parallel lines that the change of duration of the gap (expressed on a logarithmic scale) as a function of frequency is similar for all "notch" levels. That is, the rate of change of the log duration of the gap with increasing frequency is independent of the "depth" of the perceptual notch.

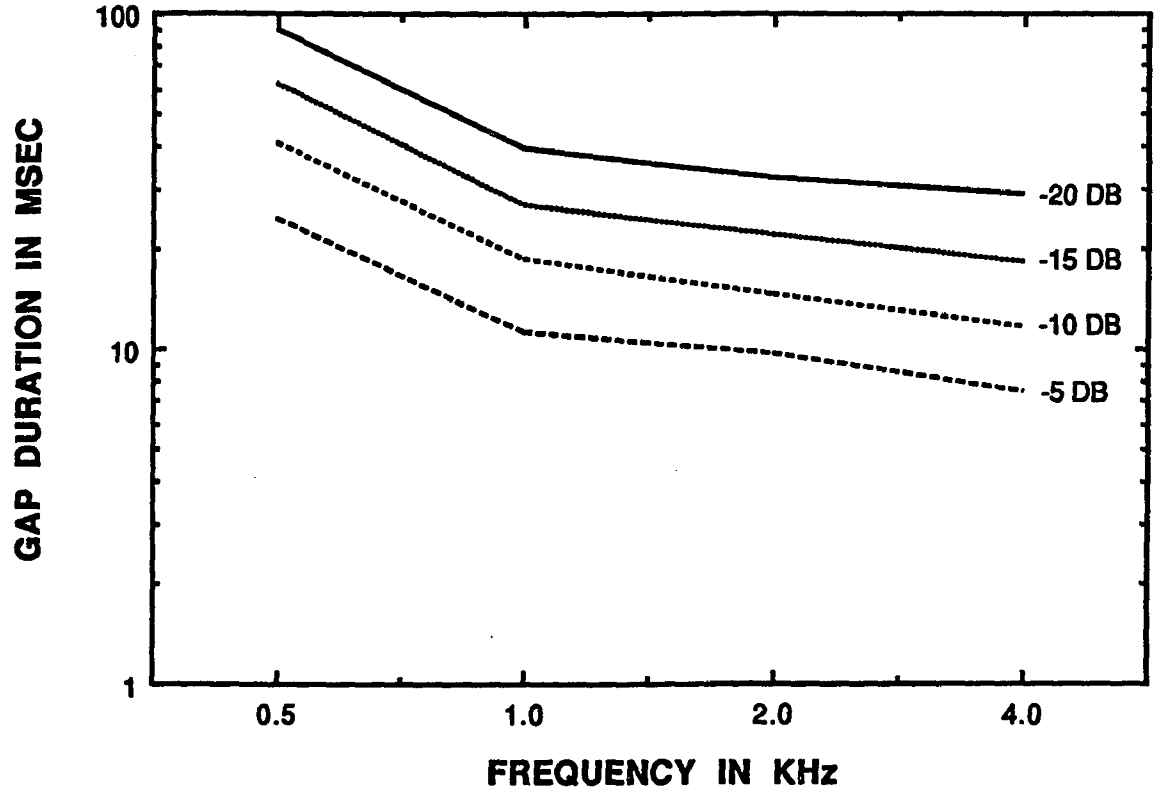

Figure 4.5. Gap Width (in msec) Expressed on a Logarithmic Scale Required to Detect Tones in Notch Depths of 5 to 20 dB as a Function of Frequency. Data are Mean of 5 Subjects.

Table 4.12 shows the results of both a repeated-measures and a simple 3-way analysis of the variance in the logarithmically transformed data shown in Table 4.7. The effects of frequency and notch depth remained highly significant ($\underline{F}$ (3,12) = 57.94, $\underline{p}$<0.01, and $\underline{F}$ (3,12) = 134.44, $\underline{p}$<0.01, respectively). Contrary to the previous findings, however, the interaction between frequency and notch depth was not found to be significant, whereas, the interaction between frequency and subject was significant ($\underline{F}$ (12.36) = 2.89, $\underline{p}$<0.01).

The Student Newman-Keuls procedure was applied to determine the differences among the means (in log duration) for the individual frequencies. The results, as shown in Table 4.13, indicate that gap widths (in log msec) for 0.5 KHz, are significantly greater ($\underline{p}$<0.01) than for frequencies 1.0, 2.0 and 4.0 KHz. This analysis also provides some evidence of frequency dependence between gap widths at 1.0 and 2.0 KHz ($\underline{p}$<0.1), and between 1.0 and 4.0 KHz ($\underline{p}$<0.01). No significant differences, however, were found between gap widths at 2.0 and 4.0 KHz. Thus, in spite of the logarithmic transformation of the data, the basic findings of a strong frequency dependence between 0.5 and 1.0 KHz and weaker frequency dependence at frequencies above 1.0 KHz remain unchanged.

Table 4.12

ANOVA Summary Table for the Effects of Frequency, Subject and Notch Depth. Data are Expressed as the Log of the Duration of the Gap Width.

SOURCE	df	MEAN SQUARE	F Repeated Measures	F Simple 3-Way
FREQUENCY (F)	3	1.087	57.94**	170.09**
SUBJECT (S)	4	0.185	---	28.42**
NOTCH DEPTH (N)	3	1.087	134.44**	170.09**
F x S	12	0.018	---	2.89**
F x N	9	0.001	0.18	0.18
S x N	12	0.008	---	1.25
F x S x N	36	0.006		

** Significant at $p < 0.01$

Table 4.13

Results of Student-Newman Keuls Procedure Applied to Differences Between the Means for Frequencies 0.5, 1, 2, and 4 KHz. Data are Expressed as the Log of the Duration of the Gap Width.

FREQUENCY (KHz)	FREQUENCY (KHz) 0.5	1.0	2.0	4.0
0.5	-	0.346**	0.437**	0.535**
1.0	-	-	0.091(*)	0.189**
2.0	-	-	-	0.098

(*) Significant at $p < 0.1$

** Significant at $p < 0.01$

Critical Differences:

Rank	$p < 0.05$	$p < 0.01$
2	0.0924	0.1296
3	0.1131	0.1512
4	0.1260	0.1650

The interaction between frequency and subject is illustrated in Figure 4.6. It can be seen from the figure that with the exception of subject MB, when gap widths are expressed on a logarithmic scale, performance as a function of frequency is similar across subjects. These results suggest that the rate of change of gap width duration (expressed on a logarithmic scale) with increasing frequency is relatively uniform across subjects.

Summary of results

1. When temporal acuity was measured in terms of the duration of the gap (in msec) and of the log duration of the gap required to detect tones at a fixed "notch" depth, there was evidence of frequency dependence.

2. In particular, temporal acuity improved considerably from 0.5 to 1.0 KHz, less from 1.0 to 2.0 KHz, and not at all from 2.0 to 4.0 KHz.

3. When the duration of the gap was measured in msec, the rate of change of temporal acuity with frequency was highly dependent on "notch" depth between 0.5 and 1.0 KHz. This was not the case from 1.0 KHz to

4.0 KHz where the rate of change was fairly constant (3.5 msec/octave).

4. When the duration of the gap was logarithmically transformed, the rate of change of temporal acuity with frequency was independent of the "depth" of the perceptual notch.

5. When the duration of the gap was logarithmically transformed, the rate of change of temporal acuity with frequency was similar for most subjects.

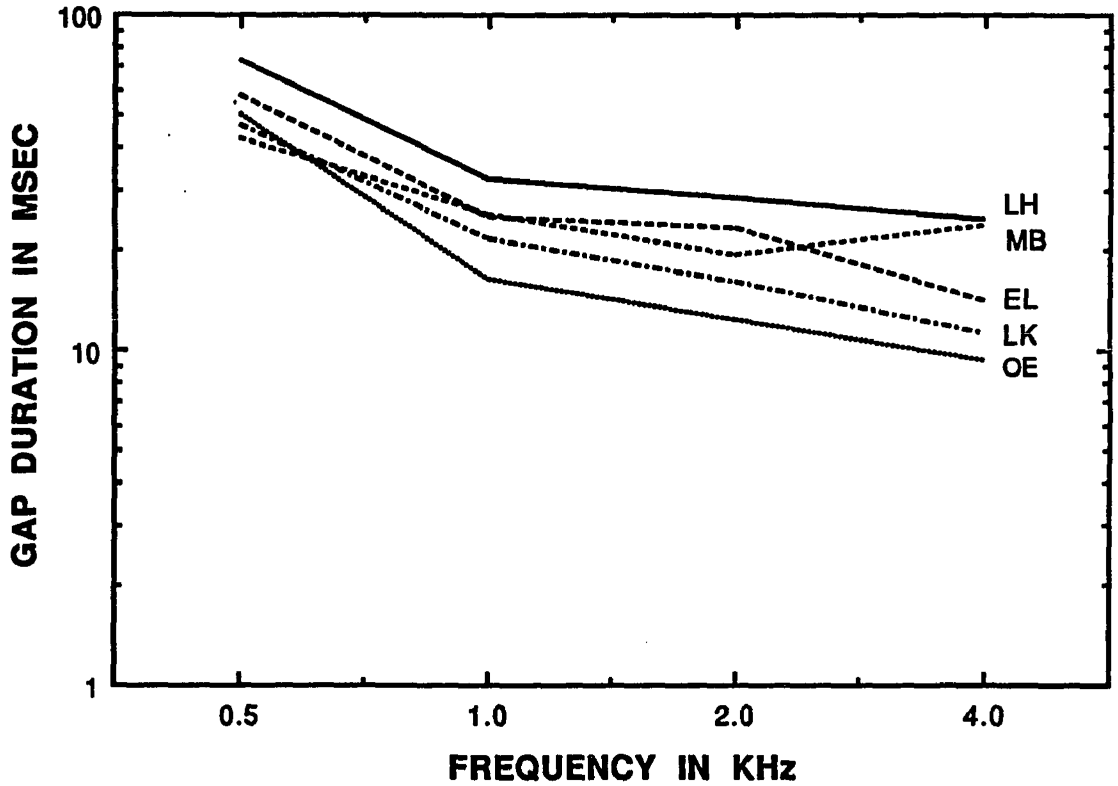

Figure 4.6. Gap Duration (in msec) Expressed on a Logarithmic Scale as a Function of Frequency for each Subject. Data are Averaged Across 4 Levels of the Perceptual Notch.

CHAPTER 5

Discussion

It will be recalled that the purpose of the present investigation was to examine temporal resolution as a function of frequency. Based on a linear filter theory, it was hypothesized that: (a) temporal resolution would improve with increasing frequency, and (b) temporal resolution would improve by a factor of approximately two for each doubling of frequency.

To test the hypotheses, an experimental paradigm was devised which, in theory, eliminates some of the procedural problems believed to have confounded data in previously published studies. These problems include: spectral splatter, time constants of external filters, "off-time" listening, and random fluctuations in the amplitude envelope of random noise. It was assumed that frequency-specific temporal resolution could be measured by examining the audibility of tones in a gap surrounded by white noise. Switching of the pure-tone occurred in the surrounding noise bands.

Experimental results were used to assess temporal resolution in two different, but related, ways: (a) by the magnitude of the tone's threshold shift in relation to it's threshold in noise for a given gap duration, and (b) by the duration of the gap (in msec) that is required to detect a tone at a given level. The first approach provides an estimate of the depth of the "perceptual notch" that is assumed to result from the brief gap in the noise. The advantage of this method is that the values of the perceptual notch depths are directly obtained from the measured data. The second method, has the advantage of providing results in time units and therefore of permitting quantitave comparisons with existing data. Its drawback is that results must be interpolated from the raw data.

Based on the original hypotheses, it was predicted that: (a) the threshold of the tone in the gap would decrease with increasing frequency for a constant gap size, and (b) the gap size would decrease with increasing frequency for a constant tone level. Furthermore, to the extent that, the gap size for a constant notch depth provided a direct estimate of temporal resolution, this gap size was expected to halve for every doubling of frequency.

Now we return to the experimental results. The data expressed in terms of the maximum notch depth at a given gap size are in keeping with the predictions that threshold improves significantly with increasing frequency, but only for short gap durations (<28 msec). For these brief gaps, significant frequency effects were found from 0.5 to 1.0 KHz, from 2.0 to 4.0 KHz, and from 1.0 to 4.0 KHz, but not from 1.0. to 2.0 KHz. Thus, at short gaps, temporal resolution expressed by the depth of the notch improved with increasing frequency as originally predicted.

For longer gaps (>28 msec), the depth of the notch increased significantly for a given gap size only from 0.5 to 1.0 KHz. Contrary to expectations, however, the depth of the perceptual notch, in absolute values, remained relatively unchanged as frequency increased from 1.0 through 4.0 KHz for a fixed gap size.

When the data were expressed in temporal units, a strong frequency effect was found from 0.5 to 1.0 KHz. Moreover, the magnitude of the gap size change at a given notch depth decreased by a factor of 2 as frequency was doubled from 0.5 to 1.0 KHz. For threshold detection in a 5 dB notch depth, for example, gap sizes decreased from 24.24 to 11.09 msec as frequency increased from 0.5 to 1.0 KHz. Thus, for low frequencies, temporal resolution was found to be inversely proportional to frequency as predicted from a linear filter theory.

For higher frequencies (> 1.0 KHz), gap durations increased significantly with frequency at a given notch depth from 1.0 to 2.0 KHz and from 1.0 to 4.0 KHz, but not from 2.0 to 4.0 KHz. These results are in keeping with the general prediction. The magnitude of gap duration change with increasing frequency at a given notch depth, however, is far less than the change predicted from the linear filter theory.

The finding of strong frequency dependence below 1.0 KHz has been previously reported by Shailer and Moore (1983). In one of the few studies that attempted to relate the nominal value of the auditory filter bandwidth to temporal resolution, Shailer and Moore found strong correlation ($\underline{r}$ = 0.995) between gap thresholds in band-limited noise supplemented with a low-level notched-noise masker and the width of the auditory filters for frequencies 0.4 and 1.0 KHz. Furthermore, for these low frequencies, gap thresholds were found to be inversely proportional to frequency as predicted from the linear filter theory.

The finding of weak evidence of frequency dependence in the data above 1.0 KHz is also in keeping with previously published results. Shailer and Moore, in their 1983 study, found that at frequencies greater than 1.0 KHz, gap thresholds continued to improve, but substantially less than predicted. They concluded, that at frequencies below 1.0 KHz, gap thresholds are limited by the "ringing" at the output of the auditory filter. At higher frequencies, there appears to be some other mechanism which limits temporal resolution. The similarity between the results reported here and those of Shailer and Moore (1983) are of particular interest. In the present study, procedural issues such as spectral splatter, external filters and amplitude envelope cues have been eliminated. Thus, the presence of these factors in the Shailer and Moore (1983) study did not appear to compromise their results.

In summary, the results of the present study showed strong frequency dependence from 0.5 to 1.0 KHz. Furthermore, the magnitude of the change was found to be inversely proportional to frequency as predicted from the linear filter theory. For higher frequencies, there was some evidence of improved temporal resolution with increasing frequency. That is, increased notch depth for a given gap size, or alternatively, decreased gap size for a given notch depth, were observed with increasing frequency. The magnitude of the change, however, was less than predicted from a simple linear filter. These results are in keeping with the published data of Shailer and Moore (1983). Moreover, they provide support for the conclusion that there is more than one mechanism limiting temporal resolution in the auditory system.

Modelling and Re-analysis of Data

For the purpose of testing the original hypotheses, the relationship between frequency and temporal resolution was measured in two ways: (a) as the depth of the perceptual notch in a given physical gap, and (b) as the duration of the physical gap required for threshold detection in a given perceptual notch. One of the more interesting aspects of the present data, however, is the relationship between the magnitude of threshold and gap duration for a given frequency. Moreover, this relationship may provide clues to the nature of the mechanisms that are responsible for limiting temporal resolution.

1. Phenomenological Description

The curves in Figure 4.1, showing threshold as a function of gap duration, are similar. Each can be described by three segments. The first segment is horizontal. That is, no change in threshold is observed relative to the threshold in noise, until a minimal gap size is reached. The second segment is steeply sloping and shows a fast decrease in threshold with increasing gap size for brief gaps only. The third segment is gradually decaying and shows a slow decrease in threshold with increasing gap size for longer gap durations. Moreover, this last segment appears to show an exponential relationship between thresholds and gap duration, the asymptote being the threshold in quiet.

2. Theoretical Physiological Interpretation

The three segments by which the data in Figure 4.1 were described may be explained by three physiological processes.

a) Stage one

The first process, where no threshold change is observed, may reflect the integration time required to sum the neural information in order to reach acceptance criterion for detection (Zwislocki, 1960; Penner and Schiffrin, 1980; Divenyi and Shannon, 1983). It is hypothesized that there is a device, also referred to as the temporal integrator, in which the process of integration occurs. Because the neural activity is inherently "noisy", the temporal integrator is assumed to have a mechanism that smoothes or integrates information over a critical period of time or "temporal window". It is possible that this first segment reflects this critical integration time.

b) Stage two

The second process, where the fast change in threshold is observed, may reflect the persistence of sound energy in the cochlea following cessation of sound input, and its rate of dissipation. This process could be thought of in two ways. One way would be analogous to room reverberation, in which energy persists because of reflections, but is dissipated because of energy absorption. Another way would be as "ringing" in the auditory filters. These may be, in fact, two different ways of looking at the same thing.

In either case, one would expect the time constant of the energy decay to fall with increasing frequency. If indeed this process reflects limitations imposed by the impulse response of the auditory filter then the time constant of this decay is expected to decrease by a factor of approximately 2 as frequency is doubled. Note that the function relating energy in the cochlea to time, would be expected to be exponential (e.g. Duifhuis, 1973; Green, 1985). However, when threshold is expressed in decibels, it should become a linear function of time. The shift from an exponential to a linear relationship as threshold is transformed from energy units to decibel units is explained in Appendix J.

If indeed there are several temporal processes which limit temporal resolution, this second process is directly related to the original hypotheses. Assuming that this process reflects the first-stage filtering at the periphery of the auditory system, then any quantitave derivations regarding the time constant for each frequency, may allow testing of the hypothesis that temporal resolution, as measured at this processing level, can be predicted from a linear auditory filter.

c) Stage three

The third segment, where a more gradual change in threshold is observed, may reflect recovery from perstimulatory adaptation. It is assumed that, in response to stimulation by the noise, there is depletion of chemical substances such as neural-transmitters with a consequent elevation in threshold. Thus, a period of time is required to recover these losses before the system returns to its pre-stimulation operating level. This process is assumed to occur at cochlear or retrocochlear levels where energy is non-linearly coded. There are behavioral data to suggest that intensity is coded logarithmically in the auditory system (Fletcher, 1938). It is, therefore, assumed that at this level, the auditory system functions on a transformed scale, approximating a log scale. Thus, threshold change reflecting recovery from adaptation is expected to be exponential on a log scale (i.e., when threshold is expressed in dB).

Consideration was also given to the possibility that stage three might represent temporal threshold integration. After all, if the duration of the tone were changed by simply varying its "on-time" rather than by placing it between noise bursts, one could obtain an exponential relationship between threshold and duration. Classical studies of temporal integration have shown the tone threshold to decay by 3 dB for each doubling of duration until a total tone duration of 200-300 msec is reached after which no further threshold improvements are observed (e.g. Plomp and Bouman, 1959; Zwislocki, 1960). The magnitude of the threshold shifts, however, are very different in the present study from those reported for temporal threshold integration in the literature. Results reported by Zwislocki (1960), for example, would predict that for a gap width of 17 msec, threshold of audibility for a 1.0 KHz tone would be only 10 dB poorer than that found for an infinitely long tone. In the present investigation, the group mean threshold for a 1.0 KHz tone in a 17 msec gap was 33 dB poorer than in an infinitely long gap. Therefore, while the possibility that temporal integration contributes to the stage three phenomenon observed in the present study cannot be eliminated, it certainly does not explain the magnitude of the effect.

3. Mathematical Modelling

Based on general morphology and the tentative explanations just introduced, the data relating threshold to gap duration were modelled by three processes. Each process is represented by a function whose form has a plausible physiological basis. Thus, this is not a purely mathematical model in which a best fitting polynomial function was derived. Rather, this model has underlying physiological explanations, based on known phenomena, potentially making it more useful in general applications.

The three processes by which the data can be explained are illustrated in Figure 5.1 by the three stages. The horizontal segment in the first stage represents the "perceptual integration time" during which the gap size is less than that required for tone threshold to begin to change from it's masked value in noise. This integration time is represented by T_1.

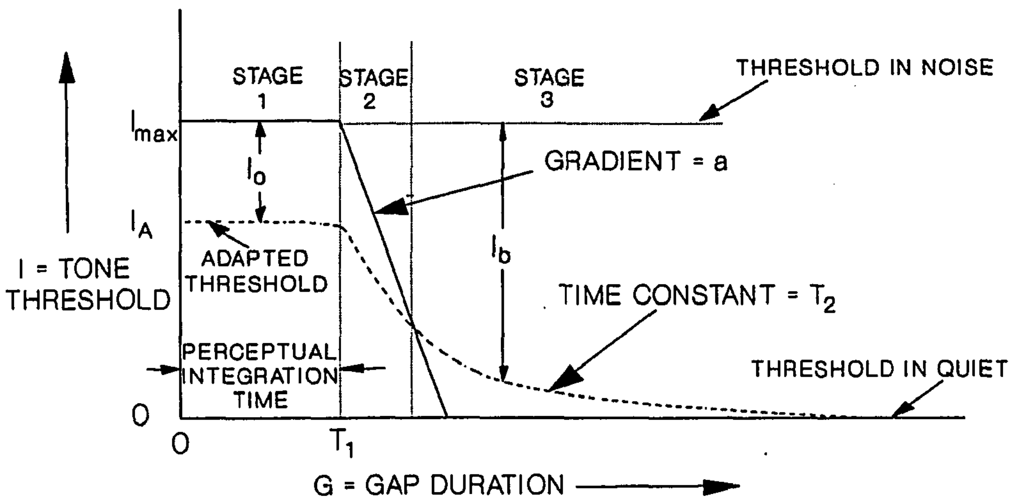

Figure 5.1. Schematic representation of the three stages of the function relating tone threshold to gap duration.

<u>Explanatory Note</u>

Stage 1: $I = I_{max}$

Stage 2: $I = I_{max} - a * (G - T_1)$

Stage 3: $I = I_A * e^{(-(G - T1)/T2)}$

Measured parameter I_b = Threshold in dB re threshold in noise = the "depth" of the perceptual notch

$I_b = I_{max} - I$
$I_o = I_{max} - I_A$ therefore, $I_A = I_{max} - I_o$

Thus, for:
Stage 1, $I_{b(1)} = 0$

Stage 2, $I_{b(2)} = a * (G - T_1)$

Stage 3, $I_{b(3)} = I_{max} - (I_{max} - I_o) * e^{(-(G-T1)/T2)}$

(see text for detailed explanation)

The linear portion in the second stage represents the decay of the internal noise energy in the cochlea following cessation of noise input. The general form of a linear function is,

$$Y = a * X + b \quad \text{.....(5.1)}$$

where a = the gradient, and b = the ordinate intercept. Thus, the attenuation of the tone in the linear segment can be described by

$$I_{b(2)} = a * (G - T_1) \quad \text{.....(5.2)}$$

where $I_{b(2)}$ = change in tone's threshold relative to it's value in noise, i.e., the depth of the perceptual notch in the second stage, a = gradient in dB/msec, G = gap duration (in msec), and T_1 = perceptual integration time (in msec). In this case, the attenuation of the tone is zero at time $G = T_1$ after which, the linear decay begins. Note also that this relationship does not hold for values of G less than T_1, where I_a = constant = zero.

The exponential decay represents the recovery from adaptation. The general equation of an exponential function is:

$$y = A * e^{-t/T} \quad \text{(5.3)}$$

where y = value of variable at time t, A = value of variable at time zero (i.e., starting value), e = base of natural logarithms (=2.718), t = elapsed time, and T = the time constant. The third stage in Figure 5.1 can be described by an exponential, where,

$$y = I_{max} - I_{b(3)} \quad \text{(5.4)}$$

where I_{max} = threshold in noise and $I_{b(3)}$ = change in tone's threshold relative to it's value in noise, i.e., the depth of the perceptual notch in the third stage. Also in the present study,

$$A = I_{max} - I_o \quad \text{(5.5)}$$

where I_o = the adapted threshold relative to the threshold in noise. And,

$$t = G - T_1 \quad \text{(5.6)}$$

where G = gap duration (in msec), and T_1 = perceptual integration time (in msec). Note again that recovery of threshold is assumed to start at time T_1, not time zero.

By substituting Equations 5.4, 5.5, and 5.6 into Equation 5.3, then

$$I_{max} - I_{b(3)} = (I_{max} - I_o) * e^{(-(G - T1)/T2)} \tag{5.7}$$

where T_2 = time constant (in msec). By rearranging Equation 5.7, then

$$I_{b(3)} = I_{max} - (I_{max} - I_o) * e^{(-(G - T1)/T2)} \tag{5.8}$$

It is clear from Figure 5.1 that, after time T_1, the measured threshold is the higher of the two thresholds predicted by Equations 5.2 and 5.8 for the linear and exponential segments, respectively. That is, for small gap durations, the decay of the internal noise in the cochlea will dominate, whereas, for longer gaps, the recovery from adapted threshold will dominate. A convenient way of deriving a function whose value is the lesser of two unrelated functions is through the use of logarithms. Thus,

$$I_b = - \text{Ln}\ (e^{-Ib(2)} + e^{-Ib(3)}) \ \ldots\ldots\ldots\ldots\ldots (5.9)$$

where I_b = threshold of tone relative to it's value in noise (in dB attenuation), Ln = natural logarithms, e = base of natural logarithms, $I_{b(2)}$ = "depth" of the perceptual notch in the second stage as determined by the decaying noise and $I_{b(3)}$ = "depth" of the perceptual notch as determined by adaptation.

The complete formula for predicting the threshold change of the tone relative to it's threshold in noise as a function of gap duration is obtained by substituting Equations 5.2 and 5.8 in Equation 5.9:

$$I_b = -Ln(e^{-a*(G-T1)} + e^{-(Imax-(Imax-Io)*e\wedge(-(G-T1)/T2))}) \tag{5.10}$$

Note that I_b, I_{max}, and I_o represent attenuation in dB and that this formula applies only for $G>T_1$.

In summary, the model, represented by Equation 5.10, predicts the threshold of the tone relative to its threshold in noise as a function of gap duration assuming that there is (a) a minimal integration time only after which the tone can detected, (b) a fast linear decay (on a dB scale) reflecting true masking due to the decay of noise energy in the cochlea after the cessation of the noise, and (c) an exponential decay possibly reflecting recovery from perstimulatory adaptation. Thus from the equation, four parameters can be extracted: minimal integration time, T_1 (in msec); the rate of the fast linear decay, a (in dB/msec); the time constant of the exponential decay, T_2 (in msec); and, the adapted threshold, I_o (in dB). The model predicts that for shorter gap durations, the linear portion will dominate, whereas, for longer gap durations the exponential portion will dominate. For both linear and exponential functions, threshold change is assumed to begin at time T_1.

4. Re-analysis of the Data using the Model

For the purpose of determining the values of the parameters T_1, T_2, a, and I_o, equation 5.10 was fitted to the group mean data using a least sum of squares procedure. For each frequency separately, this fitting process was conducted in three stages: early data for T_1 and gradient a, late data to obtain T_2 and I_o and, finally, minor adjustments to further reduce the sum of squares. Table 5.1 shows the experimental data and the predicted thresholds from Equation 5.10. At the bottom of the Table are the parameters' values that provided the least sum of squares. It can be seen that Equation 5.10 provides a good fit for the empirical data (S.D. of error of prediction = 0.53 dB). Figure 5.2. shows, for each frequency, the predicted relationship between threshold and gap duration, obtained from Equation 5.10. Also shown, by symbols, are the group means of the empirical data.

To examine the significance of frequency effects on the four parameters, a best fit curve was obtained for each subject's data, and an analysis of variance was subsequently performed for each of the four parameters of Equation 5.10. Using the same least sum of squares fitting procedure, values for parameters a, T_1, T_2, and I_o were obtained for each subject. These values, as a function of subject and frequency, are summarized in Table 5.2. Also shown in the table are the group means and standard deviations. In Appendix K are illustrations of measured and predicted data and a summary table with the least sum of squares and the standard deviation of prediction of error for each subject.

Table 5.1

Mean Tone Threshold (in dB attenuation) as a Function of Gap Width and Frequency of the Experimental Data (Exp) and the Predicted Values (Pred) from Equation 5.10 for Gap Widths 28 to 114 msec. The Data at the Bottom of the Table are Parameters for the Best Fitting Curves using the Least Sum of Squares Procedure.

GAP msec	FREQUENCY (KHz) 0.5		1.0		2.0		4.0	
	Exp	Pred	Exp	Pred	Exp	Pred	Exp	Pred
6.5	0.86	-0.05	0.66	0.54	1.56	1.51	4.10	3.71
17.2	2.98	3.17	9.55	9.44	13.35	13.37	17.16	16.96
28.0	5.92	6.39	16.96	16.90	18.10	18.42	20.53	20.58
39.0	9.20	9.60	20.11	20.43	21.58	21.32	22.38	23.60
49.5	12.82	12.75	23.32	23.30	24.28	23.90	27.22	26.26
71.0	18.65	18.11	28.33	28.03	28.29	28.23	31.49	30.66
92.5	20.46	21.21	31.84	31.66	31.69	31.65	34.74	34.06
114.0	23.71	23.42	34.34	34.46	33.84	34.36	35.34	36.70
a		0.30		0.83		1.12		1.40
T_1		6.60		5.80		5.10		3.80
T_2		96.00		82.00		91.50		83.90
I_o		5.40		8.90		11.00		12.20
I_{max}	32.17		43.79		44.58		45.71	

Sum of Squares of Error of Prediction = 8.67 dB
SD of Error = 0.53 dB
95% Confidence Limits = +/- 1.06 dB

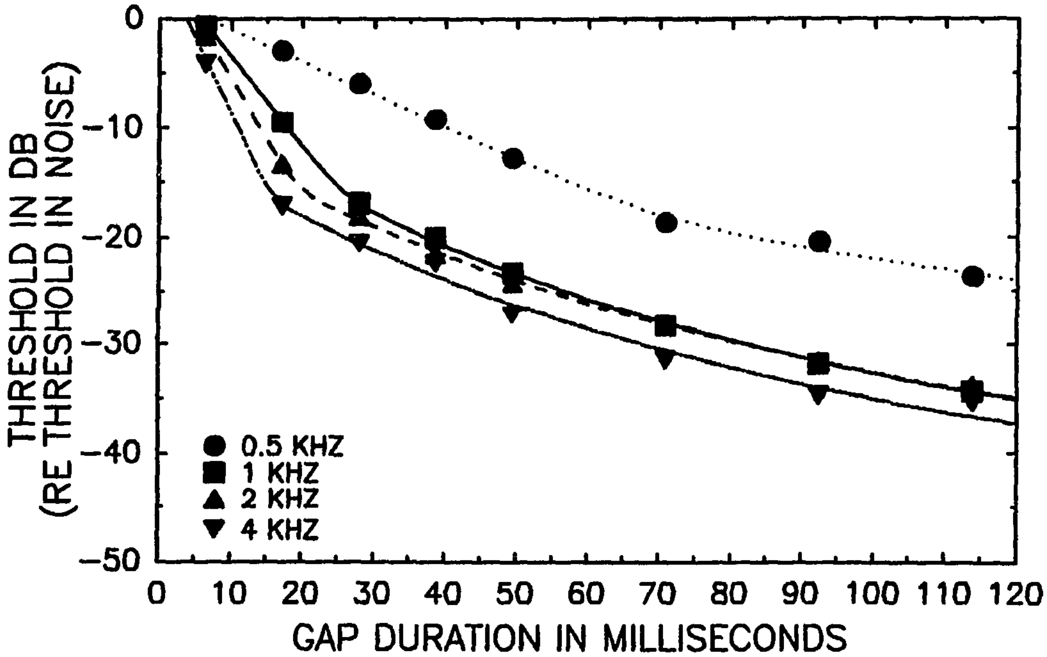

FIGURE 5.2 GROUP MEAN DATA AND BEST FITTING CURVES FROM EQUATION 5.10

Table 5.2
Summary of the Parameters in Equation 5.10 that Best Fit Individual Data using the Least Sum of Squares Procedure for Frequencies 0.5 to 4.0 KHz.

Parameter /Subject	Frequency (KHz) 0.5	1.0	2.0	4.0
a				
LK	0.3	0.8	1.3	1.5
EL	0.4	0.9	0.8	1.8
MB	0.4	0.8	1.3	0.9
OE	0.6	1.2	1.5	1.6
LH	0.2	0.5	0.9	0.8
Mean	0.38	0.84	1.16	1.32
(SD)	(0.15)	(0.25)	(0.30)	(0.44)
T_1				
LK	10.3	6.0	5.7	2.8
EL	11.1	8.0	3.2	5.6
MB	8.7	6.7	5.8	4.6
OE	2.2	5.8	3.3	1.6
LH	7.0	-1.7	8.3	1.7
Mean	7.86	4.96	5.26	3.26
(SD)	(3.53)	(3.82)	(2.11)	(1.78)
T_2				
LK	94.5	74.0	66.0	76.5
EL	134.0	85.5	79.5	76.0
MB	116.0	85.0	90.0	78.0
OE	88.0	86.0	143.0	103.0
LH	60.0	83.0	97.0	116.0
Mean	98.50	82.70	95.10	89.90
(SD)	(28.17)	(4.99)	(29.22)	(18.49)
I_o				
LK	9.3	13.0	14.8	14.5
EL	13.5	10.2	6.6	13.0
MB	13.0	6.4	10.3	6.9
OE	2.3	14.2	18.7	18.8
LH	9.0	1.0	5.0	11.0
Mean	9.42	8.96	11.08	12.84
(SD)	(4.48)	(5.37)	(5.69)	(4.39)

Because the focus of the study reported here is on temporal resolution, the gradient a, expressed in dB/msec, was converted to a time constant, expressed in msec, using the following formula:

$$a = 10 * \log(e)/T_3 \quad \text{..........................} (5.11)$$

where a = gradient of the linear segment (in dB/msec), 10 * log(e) = 4.347, conversion of energy to a dB scale, and T_3 = time constant, i.e., the time taken for energy to fall to 1/e of its starting value (in msec). For more details on the derivation of Equation 5.11 see Appendix J. Table 5.3 shows the values of the gradient, a, re-expressed as time constant T_3 for each subject as a function of frequency.

To test the significance of the frequency effect, a one-way repeated-measures analysis of variance was performed on the individual data for each parameter. A summary of the analyses of variance for parameters a (expressed as time constant T_3), T_1, T_2, and I_o, is shown in Table 5.4.

Table 5.3

Values of Parameter a Re-expressed as Time Constant, T_3, using Equation 5.11 for each Subject as a Function of Frequency.

Subject	Frequency (KHz)			
	0.5	1.0	2.0	4.0
T_3				
LK	14.48	5.43	3.34	2.90
EL	10.86	4.83	5.43	2.41
MB	10.86	5.43	3.34	4.83
OE	7.24	3.62	2.90	2.71
LH	21.71	8.69	4.83	5.43
Mean	13.03	5.60	3.97	3.66
(SD)	(5.49)	(1.88)	(1.01)	(1.37)

The results show a significant frequency effect only for the parameter T_3. Results of the Student Newman-Keuls procedure applied to the means of T_3 show that the time constant for 0.5 KHz is significantly greater than that for 1.0, 2.0, and 4.0 KHz at the 0.01 level ($q=4.58$, 5.34, and 5.83, respectively). No significant differences were found between 1.0, 2.0, and 4.0 KHz at the 0.05 level. The mean values, however, did decrease monotonically with increasing frequency from 1.0 to 4.0 KHz.

Parameters T_1, T_2, and I_0 did not vary significantly with frequency at the 0.05 level. Note, however, that values for parameter T_1 decreased monotonically with increasing frequency, and that this effect was found to be significant at the 0.1 level.

Table 5.4

Summary of One-Way Repeated-Measures ANOVAs Performed in the Data of Individual Parameters Testing the Effect of Frequency.

PARAMETER	SOURCE OF VARIANCE	SUM OF SQUARES	DEGREES OF FREEDOM	ESTIMATED MEAN SQUARE	F RATIO
T_3	frequency	289.7	3	96.56	16.94**
	error term	68.4	12	5.70	
T_1	frequency	54.1	3	18.05	2.32(*)
	error term	93.2	12	7.76	
T_2	frequency	709.8	3	236.58	0.44
	error term	6523.5	12	543.62	
I_o	frequency	46.6	3	15.55	0.69
	error term	271.5	12	22.62	

** $p<0.01$
(*) $0.05<p<0.1$

5. Discussion of Model

The experimental data relating tone threshold to gap width are well described by a mathematical model (Equation 5.10) that includes four parameters: a minimal integration time, the time constant of an early fast decaying portion, an adapted threshold, and, the time constant of the decay of a slow-changing portion. Moreover, the functions can be explained by known physiological processes. These include: energy or informational integration, cochlear energy dissipation, and recovery from perstimulatory adaptation. These processes will be discussed in turn.

(a) Perceptual Integration Time

The perceptual integration time is considered the time required to reach criterion for audibility of the test tone. It is not clear whether the phenomenon reflects integration of energy (e.g., Garner and Miller, 1947; Robinson, 1974), integration of neural excitation (Munson, 1947; Zwislocki, 1969), the time taken to achieve phase locking, or some other process. Perhaps it is a combination of these. Zwislocki's (1960) theory of temporal integration predicts that for high frequency signals, shorter durations are necessary to reach threshold of audibility than for low frequencies. This hypothesis is based on the assumption that each cycle of sinusoidal vibration of the cochlea is a single stimulus event. Thus, as the frequency increases, the time between events decreases, so that more neural activity is generated in those auditory neurons that are capable of temporal summation. Assuming that the absolute threshold is achieved when total neuron activity just exceeds some critical value, it would be predicted that psychophysical thresholds would decrease as stimulus frequency rises because of increased temporal summation. This theory has been supported by empirical data (e.g. Plomp and Bouman, 1959; Zwislocki, 1960; Watson and Gengel, 1969; Pederson and Elberling, 1972). Thus, the finding that, the values for parameter T_1 monotonically decrease with increasing frequency, are in keeping with the predictions from temporal integration theory and the published data. Note, that although there was not a statistically significant frequency effect in the values of T_1, the data do not discount such an effect because of small number of measurements and high error variance.

The absolute values for the parameter T_1, as derived here, are also in keeping with empirical data obtained in studies investigating the critical masking interval (Penner, Robinson, and Green, 1972; Penner and Cudahy, 1973). The idea of critical masking interval was introduced as a temporal analog of the critical bandwidth measure of frequency selectivity. To measure the critical masking interval, threshold for a filtered click in the temporal center of a noise masker was measured as a function of masker duration. Beyond the critical masking interval, further increase in masker duration had little effect on threshold. This critical masking interval was also thought of as a crude measure of the effective duration of an integrating device.

Penner and Cudahy (1973) found the critical masking interval to be frequency dependent. They reported critical intervals of 3.5, 5, and 10 msec for filtered clicks with bandpass of 10-5.0 KHz, 3.0-3.5 KHz and 0.1-0.6 KHz, respectively. These data are in excellent agreement with the derived temporal integration values from Equation 5.10. (see Table 5.1 for details).

Studies investigating the effect of frequency on the shape of the temporal window have also reported results that agree with the values of T_1 (Moore et al, 1988; Plack and Moore, 1990). Moore and his colleagues measured the threshold of a tone in a gap surrounded by noise maskers as a function of gap width and the temporal position of the tone in the gap. By applying the data to an intensity-weighting function, they have found that the duration of the temporal window decreases from 13 to 9 msec as the center frequency increases from 0.3 to 0.9 KHz, but continues to decrease only slightly, to 7 msec, as the center frequency increase to 8.1 KHz.

In summary, the finding that the values of T_1 decrease monotonically with increasing frequency are in keeping with both temporal summation theory and results of studies investigating the minimal integration time. Moreover, the absolute values derived for T_1 are in excellent agreement with the published data on this issue. These findings provide strong support for the hypothesis that T_1 represents a perceptual integration time. Note, however, that statistically, the main effect of frequency did not reach the 0.05 level of significance. Additional studies with more subjects and the collection of more detailed information for short intervals will be required to examine the frequency effect more precisely.

(b) Cochlear energy dissipation/ "ringing"

The values of the parameter T_3 were found to vary significantly with frequency. Specifically, the results showed a strong frequency dependence from 0.5 to 1.0 KHz, and a weak one above 1.0 KHz.

One interpretation of the data is that parameter T_3 reflects the time constant of the decay of the "ringing" at the output of the auditory filter. If this is the case, the time constant should decrease by a factor of approximately 2 for each doubling of frequency. The derived values of T_3 are roughly in keeping with this expectation for frequencies 0.5 through 2.0 KHz. Specifically, T_3 decreased by a factor of 2.3 as frequency doubled from 0.5 to 1.0 KHz, and decreased further by a factor of 1.5 as frequency increased to 2.0 KHz.

The finding that at the high frequencies the time constant for the decay of the "ringing" is larger than predicted from the linear filter theory is in keeping with published data (e.g. Shailer and Moore, 1983). It is possible that, for the high frequencies, the decay of the ringing is so brief that other processes with longer time constants dominate.

To further support the hypothesis that the parameter T_3 represents the time constant of the impulse response of the auditory filter, the values of T_3 were compared to the time constants derived from the estimated widths of the auditory filters. For the present purpose, the width of the auditory filter was approximated by the critical bandwidth. It has been shown that the critical bandwidth corresponds to a nearly constant distance of 1.3 mm along the basilar membrane, each segment innervated by 1300 neurons. Because frequency is assumed to be distributed logarithmically on the basilar membrane, the critical bandwidth increases with increasing frequency. Based on this information, Gulick, Gescheider and Frisina (1989, pg.226) approximated the critical bandwidth to be 100, 150, 300 and 800 Hz for 0.5, 1.0, 2.0 and 4.0 KHz, respectively. Assuming the time constants to be approximated by the reciprocal of filter bandwidth, the predicted values are approximately 10.0, 6.7, 3.3 and 1.3 msec for 0.5, 1.0, 2.0, and 4.0 KHz, respectively. Except for 4.0 KHz, these values are similar to the measured values in the present study, providing further support to the hypothesis that T_3 can be explained by processes relating to the auditory filter. Note that, there are many methods which approximate the critical bandwidth (e.g. Scharf, 1970). The method that was used here was chosen because of physiological correlates. Generally, the different methods provide similar estimates of the critical bandwidths, except for low frequencies (Fastl and Schorer, 1986).

To ensure that the values of T_3 did not reflect "ringing" in the headphones (TDH-49), measurements were made of the headphone's transient response to an intense click. The results showed a decay of 23 dB in 3 msec which is equivalent to a decay with a time constant of approximately 0.5 msec. This "ringing" effect (of the headphones) was considerably less than that found in the data. It must, therefore, be concluded that the experimental findings for T_3 represent transient response of the cochlea, rather than of the test equipment.

A limitation of the present study is that very few data were collected at short gap durations. At the higher frequencies, two points were available from which to estimate T_1 and T_3. The quantitave data at 2.0 and 4.0 KHz must, therefore, be considered sketchy and imprecise until more detailed measurements are available.

In summary, the values derived for T_3, are in keeping with the predictions from a linear auditory filter for frequencies 0.5, 1.0 and 2.0 KHz. Moreover, the absolute values of T_3 agree with the time constants derived from the width of the auditory filters approximated by the critical bandwidth. These effects are not the result of "ringing" in the headphones. Thus, the parameter T_3, most likely represents the time constant of the impulse response of the auditory filter as originally hypothesized. Additional data should be obtained at short gap durations, however, before drawing any further conclusions.

(c) Recovery from adaptation

The finding that the data for longer gap durations can be predicted well by an exponential function and that the time constant of this function, T_2, does not vary with frequency are in keeping with the hypothesis that this parameter represents the time constant for recovery from adaptation at a neural level. Recovery from adaptation has been investigated both physiologically and behaviorally. Physiologically, adaptation has been observed as a decline of the neural response to a constant signal until some steady level is reached at which the energy expended is just balanced by the metabolic energy available to sustain the response (Moore, 1989). When the stimulus is turned off, there is recovery of the lost chemical substances to prestimulatory levels. Physiological studies in Chinchillas, suggest that the adaptation process observed in the eighth nerve occurs in the hair cells or at the synapses between the hair cells and the eighth nerve dendrites (Abbas, 1979; Smith, 1979). In these studies, the recovery from adaptation was shown to follow an exponential function with a time constant of 35 msec.

On the issue of frequency dependence, there is no evidence that the tonotopic organization of the auditory system is accompanied by physiological differences among neurons or synapses. One would not expect, therefore, the time constant of recovery from adaptation to be different at different frequencies. Moreover, physiological data obtained in Chinchillas have not shown adaptation to be dependent on frequency (e.g., Abbas, 1979; Harris and Dallos, 1979). Thus, the absence of a frequency effect in the present study is in keeping with a neural recovery interpretation of the exponential portion of the data.

Behaviorally, adaptation has been described as the change in threshold, or loudness of a stimulus during continuous exposure (Gulick et al, 1989). This phenomenon has been commonly explored in forward masking studies. Results of these studies have shown the time constants of threshold recovery to range between 50 and 80 msec (Duifhuis, 1973; Tillman and Rosenblatt, 1975; Nelson and Freyman, 1987). These absolute values are in keeping with those reported here for T_2.

Unlike the physiological data, there are inconclusive findings on the effect of frequency on behavioral adaptation. Some studies revealed more forward masking at high frequencies (e.g. Luscher and Zwislocki, 1949; Harris and Rawnsley, 1953) whereas others showed more masking at low frequencies (e.g. Jesteadt et al, 1982). Jesteadt et al (1982) have attributed these discrepancies to procedural issues. They have concluded that forward masking is increased at the low freuencies and that this results from growth of masking that varies with frequency and not changes in the recovery process. Based on their experimental paradigm, however, these two processes could not be separated.

On the basis of the results of the present study it seems possible that the published data of forward masking studies reflect cochlear ringing and perceptual integration effects in addition to those of post-stimulatory adaptation. To the extent that two of these effects may show frequency-dependence while the other does not, it is easy to see how forward masking studies might or might not show such dependence. One of the virtues of the model developed here is that it permits the identification and separation of these three mechanisms.

It should further be noted that many of the forward masking data reported in the literature are plotted on a logarithmic time scale (e.g. Jesteadt et al, 1982; Glasberg, et al 1987). By doing this, more weight is given to the data for longer time intervals and the rate of threshold change at short durations is, therefore, obscured. In one of the few studies where forward masking data were plotted on a linear time scale (e.g. Festen and Plomp, 1983), it was clear that the data of normally-hearing listeners can be described by two stages. The first stage shows fast threshold change up to 20 msec masker-stimulus delay, whereas the second stage shows a slow threshold change. Because Festen and Plomp's (1983) forward masking functions are based on only three masker-stimulus time delays, no further conclusions can be drawn. Thus, the possibility that much of the forward masking data in the literature hides the effects of cochlear "ringing" cannot be precluded.

Thus, there is evidence from physiological and behavioral studies of recovery from adaptation to support the conclusion that the parameter T_2 represents the time constant of recovery from adaptation at a neural level. The time constant of this process was found to be 88 msec, independent of frequency.

6. Summary

In summary, the data are well fit by a mathematical function involving four parameters, each possibly reflecting the temporal constraint of a different processing level. One parameter is the time constant of the transient response of the cochlea, possibly reflecting ringing of peripheral auditory filters. For frequencies 0.5 to 2.0 KHz, the time constant is quite well predicted from the auditory filter theory. That is, there is a decrease of the time constant's duration by a factor of 2 for each doubling of frequency and the values are approximately equal to the reciprocal of critical bandwidth. The auditory filter theory does not well predict data for 4.0 KHz, but these data are imprecise because of the small number of measurements made for gaps of very brief duration. Another parameter reflects the time constant of a slow exponential process which is independent of frequency. There is evidence to support the hypothesis that this parameter represents recovery from adaptation at a neural level. The third factor is believed to reflect a perceptual integration time required to reach some criterion for audibility of the test tone. This duration decreased monotonically with increasing frequency but the frequency effect did not reach the 0.05 level of significance. The issue of frequency-dependence should be resolved by more sensitive experimental design. Note that the fourth parameter, I_o, possibly reflecting the adapted threshold, has not been discussed because it does not represent a temporal process.

Procedural Issues

It will be recalled that the experimental paradigm used in the present investigation was devised to eliminate many of the procedural problems associated with published experiments. One such problem is the issue of temporal constraints of external filters. In the present paradigm no external filters were used to limit the frequency range of the noise or the pure-tone signal. There may, however, have been other ways where time constants could have been introduced. Examples might be the transient response of the headphones and response time of the electronic switch. It has been established that the ringing of the headphones was shorter than the time constants found in the data (see previous section for more details). Moreover, the rise time of the electronic switch was set to 0.1 msec. It seems highly unlikely, therefore, that the equipment artifacts were mistaken for auditory phenomena in this study.

Another procedural problem that the present study was designed to eliminate is that of spectral splatter associated with switching a pure-tone. This problem has eliminated by initiating and ending the tone in the surrounding noise maskers. Thus, it is assumed that, at low signal levels, any switching of transients would be masked by the noise. Note, however, that when the gap size was very small, the threshold of audibility of the signal was similar to the level of the noise. Thus, it is possible that, for these brief gaps, spectral splatter was audible. This appears unlikely because listeners' reported that they were hearing and detecting a tone. Moreover, the rise/fall time of the pure-tone was 10 msec. Empirical data, however, should be provided to support these subjective impressions.

Embedding the onset and offset of the tone in the noise to avoid spectral splatter problems introduced an additional problem. It was assumed that in most conditions, with the exception of very brief gaps, the segments of the tone in the noise were inaudible. It is not clear, however, to what extent this additional energy may have contributed to the detection of the tone in the gap. Some support for the assumption that there would be no contribution of low-level energy to the decision making of the tone in the gap comes from physiological studies in chinchillas. Smith (1979) found that adding a less intense to an existing intense sound did not increase neural response. This phenomenon, however, should be examined in behavioral studies by varying the onset and offset time of the tone.

The present study was not designed to separate the effects of forward and backward masking. This issue, however, would be of relevance to the main issues in the present study if the phenomenon of backward masking is frequency dependent. It has been hypothesized that backward masking stems from interactions between probe masker and masker excitation patterns at the periphery. (Duifhuis, 1973; Dolan and Small, 1984). The auditory filter theory would predict that, for higher frequencies, less interaction (overlap) between the probe and the masker would occur and therefore less backward masking. This theory has been supported empirically (e.g. Elliott, 1962; Patterson, 1971; Duifhuis, 1973; Dolan and Small, 1984). Dolan and Small (1984), for example, found that the effects of backward masking using broadband noise decrease with increasing frequency of sinusoidal probes. When fitting the data with an exponential function for each frequency, the time constants were found to decay monotonically from 3.82 to 1.77 msec for frequencies 0.5 to 7.0 KHz. Again, it is not apparent to what extent procedural issues such as spectral splatter and "off-time" listening might have affected the results. If indeed this phenomenon is frequency dependent, the effect of backward masking might need to be considered in the proposed model (Equation 5.10).

In summary, the paradigm devised to examine frequency dependence of temporal resolution eliminated many procedural problems associated with published studies but could have introduced others. These problems include confounding with time constants of the electrical switches and headphones, detection of transients at very brief gap durations, temporal integration, and the effects of backward masking. All these issues should be further investigated.

Other relevant paradigms

Although the paradigm used in the present experiment was a novel one, there have been similar paradigms reported in the literature. Many studies have investigated the combined effect of forward and backward masking by measuring the threshold of a brief stimulus centered in a gap surrounded by noise as a function of gap duration (e.g. Raab, 1961; Pollack, 1964; Elliot, 1969; Patterson, 1971; Wilson and Carhart, 1971; Robinson and Pollack, 1971, 1973; Penner, 1980). The results of these studies have shown a combined masking effect that is larger than the effect of each masking paradigm (either forward or backward masking) when presented separately, and greater than what might be expected for the sum of dB masking of the two effects. Robinson and Pollack (1973), for example, reported an additional masking in the combined condition (re the intensity-sum of forward and backward masking) of 6-12 dB for a click centered in a gap of 20 msec. One possible explanation to these results is that "off-time" listening contributes to the audibility of the tone in the forward masking experiment. Thus, one consequence of the introduction of the backward masker is to eliminate "off-time" listening.

In theory, the concept underlying the combined effect of forward and backward masking is similar to the concept of the perceptual notch. That is, the stimulus is detected only if the signal's duration is sustained for a critical period of time above either the level of the internal noise or the recovering adapted threshold. The results of the combined masking studies suggest that the detection of tones in gaps in the present study may be affected by a combination of both forward and backward masking. Because the effects of backward masking do not extend beyond approximately 3.0 to 5.0 msec (e.g. Osman and Raab, 1963; Babkoff and

Sutton, 1968, 1970), the combined effects would be expected to be present at very brief gaps only.

Zwicker and Schorn (1982) have combined the effects of pre-stimulus, simultaneous, and post-stimulus masking in a paradigm called masking pattern periods. In this paradigm, the threshold of a brief test tone (5 msec) is measured as a function of its temporal position within the period of the masker. For example, for a masker repetition rate of 15.6 Hz, the period is 64 msec. During that period, the masker with a 1 msec rise and decay time is 32 msec on and 32 msec off. The threshold of the test tone is measured every few msec within that period. The resulting masking-period pattern presumably provides information on the temporal resolution of the ear. Specifically, Zwicker and Schorn have suggested that the relationship of the difference between the maximum and the minimum of the masking period patterns to the difference between the minimum and threshold in quiet should provide a quantitave measure of auditory temporal resolution. From data collected in normally-hearing listeners, this value approached 1. Any significant deviation from 1 suggests reduced temporal resolution as was found by Zwicker and Schorn in listeners with hearing impairment caused by presbyacusis, noise, and Meniere's disease.

Again, the underlying concept is similar to the perceptual notch depth because the threshold of the tone in the period when the masker is off would depend on the rate of decay of the internal noise. Note, however, that some of the procedural problems stated previously apply to this paradigm as well. Specifically, it is not clear whether spectral splatter contributed to the detection of the brief tone. Also, filtered noise maskers were used, therefore, introducing additional time constants.

A similar paradigm to the paradigm presented here and to that of Zwicker and Schorn has been used in investigations of the shape of the temporal window (Moore, Glasberg, Plack and Biswas, 1988; Plack and Moore, 1988). To obtain the shape of the temporal integrator, thresholds of brief tone pulses positioned in a temporal gap between two noise bursts were measured as a function of gap duration and of the relative position of the tone pulse within the gap. To avoid problems of spectral splatter associated with the brief pulse, a low-level notched-noise masker was presented as well. This paradigm allowed quantitave measurements of the "off-time" listening used by listeners. Note, however, that the paradigm is not directly comparable with the one used in the present study because of different test stimuli, the use of notched-noise maskers and the use of external filters.

Conclusions

The results of this study support the following conclusions:

1. The ability of the auditory system to respond to a new stimulus in the time immediately following the cessation of a previous stimulus is limited by at least two factors.

   The first of these factors is dominant at very short durations and most probably reflects the persistence of sound energy in the cochlea. It's time constant varies from approximately 13 msec at 0.5 KHz to approximately 3 msec at 4.0 KHz and is clearly frequency-dependent. Moreover, these time constant values are in keeping with the conclusion that this persistence of sound in the cochlea is equivalent to "ringing" in the peripheral auditory filter.

   The second factor is dominant at longer durations and most probably reflects recovery from perstimulatory adaptation at a neural level. It's time constant is approximately 90 msec and is independent of frequency.

2. In the experimental paradigm used here, an additional temporal factor was apparent that most probably represents the minimal time over which information about a test tone must be available before a criterion for detectability is reached. This time averages between 3 and 8 msec and may be frequency-dependent.

3. The existence of several sources of temporal constraint in the auditory system, and of several possible experimental paradigms for investigating them, precludes a simple definition of auditory temporal resolution - either qualitatively or quantitatively.

CHAPTER 6

FUTURE RESEARCH AND CLINICAL IMPLICATIONS

The results of the present study appear to reveal the effects of three different processes, each of which imposes a different temporal constraint on performance. It is hypothesized that these processes are:

(a) Integration of energy and/or information prior to attainment of a perceptual decision criterion;

(b) Exponential decay of sound energy within the cochlea following cessation of sound input; and,

(c) Recovery from perstimulatory adaptation.

It is suggested that follow-up work follow three lines of inquiry:

(1) Collection of more detailed data so as to more clearly describe the phenomena associated with the detection of tones within temporal gaps in noise;

(2) Testing of the hypothetical model developed to explain the current data; and,

(3) Explanation of the effects of auditory pathology on the temporal properties of the auditory system and their implications for rehabilitative intervention.

On the issue of further data collection in normally-hearing listeners, it is essential to measure performance at closer intervals of time for small gap durations (<20 msec). From the present data, conclusions about perceptual integration time and the rate of energy decay in the cochlea are based on only a few data points. This is a particular problem in the high frequencies where both perceptual integration time and the time constant of cochlear energy decay are very short.

A second issue related to data collection is that of noise intensity. The present data were collected with only one noise level. An implicit assumption of the mathematical model developed to explain the data is that the temporal properties of the auditory system as measured in the present paradigm, are independent of noise level. This assumption is clearly open to empirical test.

Thirdly, more detailed information is required in the frequency domain. Conclusions about the nature of the relationship between time constant and frequency cannot be reliably drawn on the basis of only a few frequencies.

The collection of more detailed information would make it possible to answer the following four questions related to the phenomena under investigation:

(1) Is the relationship between threshold and gap duration truly linear (in terms of dB versus time) during that portion of the curve believed to reflect the effects of cochlear energy dissipation? If the answer is yes, this would imply that the threshold improvement is exponential (in terms of energy versus time). A positive answer would also provide support for the hypothesis that the underlying mechanism is peripheral and mechanical.

(2) Are the time constants and the perceptual integration time independent of noise level? If the answer is yes, the implication would be that the underlying processes are linear. This would be expected for the peripheral mechanical process, and for the perceptual integration time (which is a function of the tone and not the noise). Non-linear effects, however, may well occur for the recovery from perstimulatory adaptation. If such effects are observed (i.e. if the long time constant, T_2, is a function of noise level), this finding will have important implications for experiments with hearing-impaired subjects. In

particular, the issue of presentation level, that is, SPL versus SL will need to be addressed when comparing normally-hearing with hearing-impaired subjects (e.g., Buus and Florentine, 1985)

(3) Is the fast time constant, T_3, inversely proportional to critical bandwidth? More definitive tests of the predictions of an auditory filter model will be possible with more reliable measures of T_3 at finer intervals of frequency.

(4) What is the relationship between perstimulatory adapted threshold (I_0 in the model of equation 5.6) and noise level? In this case non-linear effects would be expected. At low noise levels, it would be reasonable to assume that the neural and synaptic systems can keep pace with ongoing needs. The effects of chemical depletion, however, would be expected to increase rapidly, once stimulation exceeds a critical level.

As indicated in the introduction to this chapter, a second purpose of the follow-up work is to test the hypothetical model developed to explain the present data. Much of the detailed data collection just outlined will contribute to this second purpose. An additional line of investigation might involve replication of the study in cochlear implant recipients. If the proposed model is correct, then cochlear implantees should show no mechanical cochlear effects (e.g. Shannon, 1989), but their recovery from perstimulatory adaptation might be relatively normal. If the time constant for recovery from adaptation is indeed normal, this would support the conclusion that perstimulatory adaptation, as measured by this paradigm, occurs at retrocochlear levels. If, however, the normal adaptation occurs at hair cell/dendrite synapses (as suggested by Abbas, 1979; Smith, 1979), then cochlear implantees might show faster-than-normal recovery.

Experiments with cochlear implant recipients involve the use of hearing-impaired individuals with known "pathology" to test the hypothetical model. Most of the follow-up work with hearing-impaired subjects, however, will involve use of the experimental paradigm and the proposed model to provide insights into the pathologies associated with sensorineural hearing loss. Thus, a first step will be to replicate the experiments in individuals with sensorineural hearing loss. To the extent that the three temporal measures reflect different physiological mechanisms, it is expected that they will be differentially affected by sensorineural damage. It is possible, however, that one or more of the time constants will be unaffected. Also, to the extent that individuals with similar thresholds may have different underlying pathologies, it is expected that the relative effects of sensorineural hearing loss on the three temporal parameters will vary from individual to individual.

An additional line of inquiry may be the relationship between temporal resolution and speech perception in the hearing-impaired. There is limited evidence that deficits in temporal processing may be partly responsible for poor speech perception in the hearing impaired (e.g. Tyler et al, 1982; Dreschler and Plomp, 1985; Irwin and McAuley, 1987; Summerfield, 1987; Glasberg and Moore, 1989). In the published studies, the most common paradigm used to asses temporal resolution is gap detection. It is possible, that by measuring temporal resolution with a paradigm that will reflect time constraints of different temporal processes, a clearer picture will emerge regarding the correlation between speech perception and temporal resolution. Furthermore, the provision of three temporal measures may help explain why there are hearing-impaired individuals with similar hearing sensitivity but with a wide range of speech perception ability.

Successful correlations between measures of temporal resolution and speech perception may have important implications regarding the candidacy for cochlear implant. Pre-implant tests have shown limited correlation between performance using the common psychoacoustic methods and performance with the cochlear implant (e.g., Brown, Dowell, Clark, Martin and Pyman, 1983). Hochmair-Desoyer, Stiglbrunner and Wallenburg (1984), however, have reported that the improvement of hearing in recipients of the Vienna cochlear implant depended on the their temporal resolution abilities as assessed in gap detection and temporal difference limen tasks. It appears, therefore, that for specific cochlear implant devices, listeners may rely heavily on their temporal resolution ability. Thus, determining the temporal processing capabilities of cochlear implant candidates may be useful for both patient and device selection.

Another potentially valuable area of follow-up is the application of the experimental paradigm to populations having possible auditory perceptual difficulties unassociated with threshold shift. Obvious examples are the elderly and the learning-disabled. There is already evidence in the literature that temporal resolution deteriorates with aging (e.g., Price and Simon, 1984). It also has been suggested that certain types of auditory perceptual disorder are directly attributable to poor temporal resolution. A paradigm that can separate mechanical cochlear effects, perstimulatory adaptation effects, and perceptual integration effects, would be particularly useful in exploring auditory functions in these populations.

In addition to possible information about auditory pathology and its effects on speech perception, a more detailed understanding of the temporal constraints of the damaged auditory system could have important implications for the design of hearing aids and other sensory aids. With the use of modern techniques, such as digital signal processing, acoustic cues can be enhanced in the time domain (e.g., Boothroyd, 1990). It is important, however, to ensure that temporal modifications do not interact unfavorably with temporal limitations within the damaged auditory system. For example, Boothroyd, Springer, Smith and Schulman (1988) reported that performance on a feature perception test, of eight (out of nine) hearing-impaired individuals, decreased when amplitude compression was introduced.

Based on these and other similar findings, Boothroyd et al (1988) and Plomp (1988) argued that amplitude compression may remove significant time-intensity information in the speech signal, especially when the time constants of amplitude compression are small. This would be expected to be particularly serious for those hearing-impaired subjects who are known to have poor temporal resolution. There are, however, a few cases where amplitude compression has increased the discrimination of the speech signal. It is possible, that performance of these individuals with amplitude compression could be predicted from their auditory temporal capabilities as measured in the present paradigm.

In short, the opportunities for refinement, expansion, and application of this line of research are numerous.

APPENDIX A

SUBJECT INFORMATION

Table A.1

Individual subject history.

SUBJECT	SEX	AGE	EXPERIENCE IN PSYCHO-ACOUSTICS EXPERIMENTS
LK	F	28	YES
EL	M	23	NO
MB	M	40	YES
OE	F	28	NO
LH	F	32	NO

Table A.2

Hearing thresholds in dB HL of the tested ear.

SUBJECT	TEST EAR	THRESHOLD (in dB HL) 0.5	1.0	2.0 (in KHz)	4.0	8.0
LK	RIGHT	5	0	0	5	0
EL	LEFT	5	0	0	0	0
MB	RIGHT	0	0	0	5	10
OE	LEFT	5	0	5	0	0
LH	RIGHT	15	10	0	15	0

APPENDIX B

FREQUENCY MEASUREMENTS OF PROBE TONES

The frequency of the probe tone was determined by performing a Fast Fourier Transformation (FFT) on the stimulus in the gap. The FFT used here is part of a digital waveform analysis program DADISP. The expected frequency and the measured frequency of a sample of probe tones will be seen in Table B.1.

Table B.1

Frequency Measurements of a Sample of the Probe Tones.

GAP (Msec)	EXPECTED FREQUENCY (KHz) A	MEASURED FREQUENCY (KHz) B	DIFFERENCE (KHz) lB-Al	% DIFFERENCE (lB-Al)/A *100
46	0.5	0.4955	0.0045	0.9
16	1.0	1.0000	0.0000	0.0
26	1.0	1.0040	0.0040	0.4
106	1.0	0.9978	0.0022	0.2
46	2.0	2.0020	0.0020	0.1
6	4.0	3.9695	0.0305	0.8
6	4.0	4.0000	0.0000	0.0
MEAN	---	---	0.0062	0.3
S.D.	---	---	0.0109	0.4

APPENDIX C

DURATION MEASUREMENTS OF CONTROL AND TEST STIMULI

The durations of the control and test stimuli in a sample of trials were measured using a digital oscilloscope and a digital waveform analysis system developed at CUNY. A consistent difference was found between the duration of the control stimulus and that of the test stimulus as shown in Table C.1.

It was hypothesized that the difference in duration between the control and test stimuli was software related. Specifically, the computation of an additional power function in the generation process of the control stimulus (in order to set the switch that would output the probe tone) could prolong the overall duration of the stimulus. Indeed, the time required to compute a power function using the Apple II computer and a A/D output of a Cyborg ISAAC interface was measured to be in the same order of magnitude as the nominal difference in duration between the control and test stimuli.

Table C.1

Measurements of Duration of Control Stimuli and Test Stimuli

FREQUENCY (KHz)	GAP (Msec)	MEASURED DURATION (MSEC) CONTROL STIMULUS A	TEST STIMULUS B	DIFFERENCE (Msec) B-A	% DIFFERENCE (B-A)/A *100
0.5	0	791.3	814.7	23.4	2.96
0.5	16	790.9	814.2	23.3	2.95
0.5	46	786.8	812.1	25.3	3.22
1.0	16	793.0	813.0	20.0	2.52
1.0	26	789.2	814.1	24.9	3.16
1.0	106	788.2	814.0	25.8	3.27
2.0	46	791.3	813.5	22.2	2.81
2.0	66	790.6	813.8	23.2	2.93
4.0	6	791.8	814.1	22.3	2.82
4.0	6	788.5	813.5	25.0	3.17
MEAN	---	790.2	813.7	23.5	2.98
S.D.	---	1.9	0.7	1.8	0.23

APPENDIX D

GAP DURATION MEASUREMENTS

The duration of gaps in a sample of trials was measured using a digital oscilloscope and a digital waveform analysis system developed at CUNY. The measured gap widths were found to be consistently larger (an average of 7.5 %), than the expected values requested via software. The difference between the expected and measured gap durations, in msec and percent, will be seen in Table D.1. The source of the discrepancy between the expected and measured gaps is probably in the inaccurate value of the gap array's output rate reported by Applesoft. The actual output rate of the array was found to be approximately 7% to 10% longer than the 2 msec reported by Applesoft. This difference corresponds to the difference in duration between the expected and measured gaps in this study.

Table D.1

Measurements of Duration of the Gaps in Control and Test Stimuli

EXPECTED	GAP IN					
	CONTROL STIMULUS			TEST STIMULUS		
GAP (Msec) A	Measured (Msec) B	Diff. (Msec) B-A	% Diff. (B-A)/A *100	Measured (Msec) C	Diff (Msec) C-A	% Diff (C-A)/A *100
0	0.0	0.0	0.0	0.0	0.0	0.0
6	6.6	0.6	10.0	6.5	0.5	8.3
6	6.6	0.6	10.0	6.7	0.7	11.7
8	--	--	--	8.6	0.6	7.5
16	17.8	1.8	11.3	17.1	1.1	6.9
16	17.6	1.6	10.0	17.0	1.0	6.3
16	--	--	--	17.2	1.2	7.5
16	--	--	--	17.1	1.1	6.9
26	28.5	2.5	9.6	28.1	2.1	8.1
32	--	--	--	34.2	2.2	6.9
46	49.4	3.4	7.4	49.5	3.5	7.6
46	49.5	3.5	7.6	49.6	3.6	7.8
64	--	--	--	68.2	4.2	6.5
64	--	--	--	68.5	4.5	7.0
66	71.0	5.0	7.6	70.5	4.5	6.8
106	112.9	6.9	6.5	113.5	7.5	7.0
128	--	--	--	136.7	8.7	6.8
MEAN	--	--	8.0	--	--	7.0
S.D.	--	--	3.2	--	--	2.2

APPENDIX E

LINEARITY MEASUREMENTS OF MANUAL AND PROGRAMMABLE ATTENUATORS AND THE COMPLETE SYSTEM

a) <u>Linearity of manual attenuator</u>

The linearity of the manual attenuator was determined by comparing the level of an input pure-tone of 300 millivolt at the output of the attenuator to the expected level of attenuation (as determined by the nominal value on the manual attenuator). An A.C. Millivoltmeter (GSC, 1289) was used to measure the level of the tone. The attenuator performed close to expectations. Note that The measured attenuation of the tone at the output of the attenuator and the expected attenuation levels will be seen in Table E.1.

Table E.1.

Calibration Measurements of the Manual Attenuator.

Pure-Tone Frequency (KHz)	Nominal Value (dB Att)	Measured Value (dB Att)
0.5	50	47.50
1.0	50	47.50
2.0	50	47.00
4.0	50	47.50

b) <u>Linearity of programmable attenuator</u>

The linearity of the programmable attenuator was determined by comparing the level of a tone at the output of the attenuator to the expected level of attenuation. An A.C. Millivoltmeter (GSC, 1289) was used to measure the level of the tone. The results are illustrated in Figure E.2.1. It will be seen that, for all frequencies, the attenuator is perfectly linear at 60 dB attenuation or less. At attenuation levels greater than 60 dB, the measured attenuation becomes less than the expected attenuation.

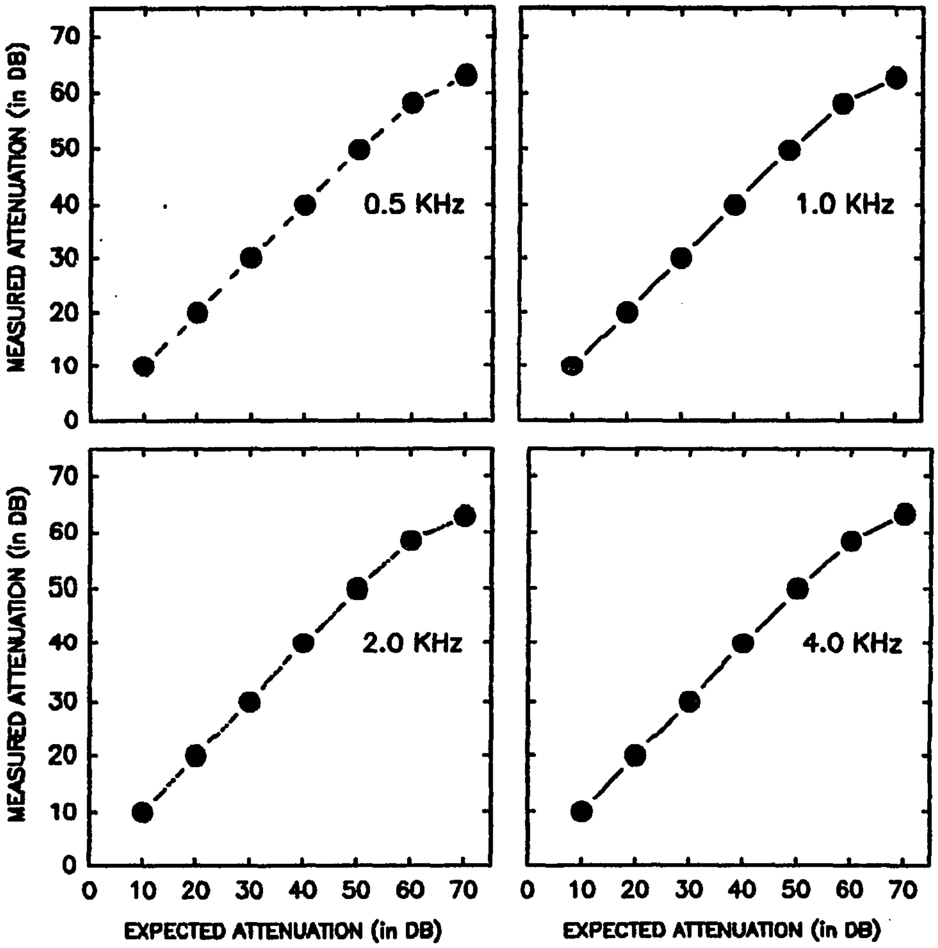

FIGURE E.1 NOMINAL VS MEASURED ATTENUATION OF PROGRAMMABLE ATTENUATOR

c) Linearity of the system

The linearity and accuracy of the complete system was determined by comparing the level of the tone at threshold as determined by the software to the level measured by a Bruel & Kjaer (B & K) Sound Level Meter (Type 1613) at the output of the headphone using a B & K Artificial Ear (6 cc coupler, Type 4152) and a B & K Microphone (Type 4144). The results are illustrated in Figures E.2 for input pure-tone frequencies of 0.5 to 4.0 KHz. It will be seen that the attenuator performed as expected for thresholds of 60 dB attenuation or less. Because the fixed attenuator was set to 20 dB, and the best threshold in quiet did not exceed 72 dB attenuation, the programmable attenuator was never greater than 52 dB attenuation. Thus, the system always operated in it's linear region.

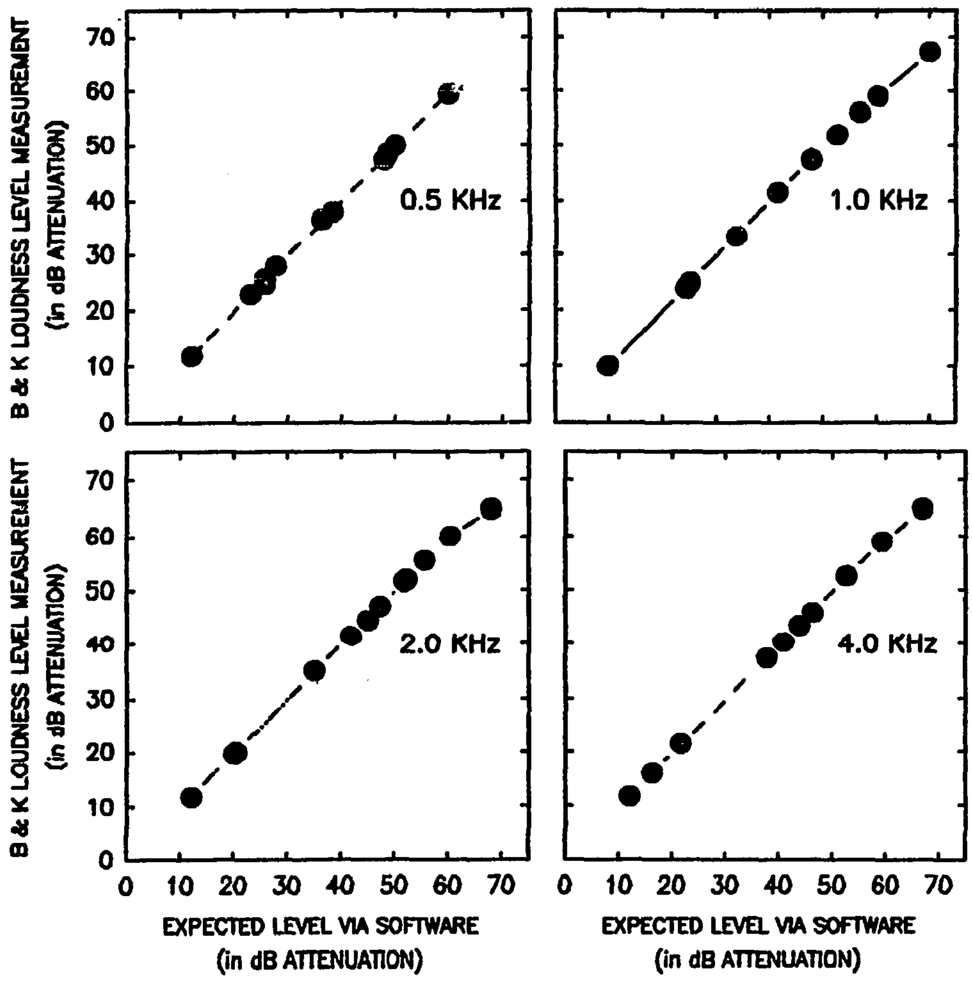

FIGURE E.2 NOMINAL VS MEASURED ATTENUATION
OF THE COMPLETE TEST SYSTEM

APPENDIX F

SAMPLE PRINT-OUT OF SUBJECT DATA

DATE OF TEST 12/14/87

SUBJECT NAME LK

GAP DURATION 4 MSEC

FREQUENCY 1000 (998) HZ

MANUAL ATTEN 50 DB

START TIME 6:56:4

END TIME 6:58:45

TIME TAKEN 2 MIN AND 41 MSEC

1: DURATION OF STIMULUS .6 SEC

2: INTERSTIMULUS INTERVAL .5 SEC

3: STIMULI PER TRIAL 2

4: HITS TO REVERSE 2

5: REVERSALS AT MIN TO QUIT 7

6: STARTING ATTENUATION 0 DB

7: STARTING STEPSIZE 16 DB

8: MINIMUM STEPSIZE 2 DB

9: TONE IN NOISE

ALL DATA

1:	0
2:	16
3:	32
4:	48
5:	40
6:	44
7:	48
8:	52
9:	50
10:	48
11:	46
12:	44
13:	42
14:	40
15:	38
16:	40
17:	38
18:	36.
19:	38
20:	36
21:	38
22:	40

REVERSALS:

38	40	36
38	36	40

MEAN: 38 DB

APPENDIX G

LISTING OF BASIC COMPUTER PROGRAM

100 - Dimension Arrays

```
100     DIM GT(100), GS(100)
        REM Dimensions for arrays of gap with tone and a
        silent gap. Each point is 2 msec, i.e., the gaps
        could be max 200 msec.
110     DIM X(20)
        REM Store attenuation at each reversal at min
        step size.
120     DIM Y(50)
```

200 - Parameters for Adaptive Procedure

```
210     R = 0
        REM Set reversals.
220     AC = 0
        REM Current attenuation level of tone.
225     AS = AC
        REM Starting attenuation.
230     SC = 16
        REM Current stepsize.
232     SS = SC
        REM SS = starting stepsize.
240     SM = 2
        REM Minimum stepsize.
250     PAD = 1
        REM ascend/descend parameter.
255     NT = 2
        REM Number of tokens per stimulus.
260     NS = 2
        REM Number of hits before further attenuating. If
        the number of hits are less than NS, then
        increase tone level.
261     REM NS=1 GIVES P=.5. NS=2 GIVES P=.71. NS=3 GIVES
        P=.79. NS=4 GIVES P=.84.
270     RQ = 10
        REM Reversals to quit at min step size.
275     NA = 0
        REM Number of times attenuator was set.
280     & BOUT, (DV) = 0
        REM Turns off noise, tones, etc.
290     DN = .6
        REM Noise duration in sec. Here it is .6 sec,
        i.e., each burst is 300 msec.
```

```
300    DI = 0.5
       REM Interstimulus time.
310    BN = 8
       REM The binary out number for the noise is 8
       since the tone attenuation occupies binary out
       numbers 0 to 7.
320    BT = 9
       REM The binary out number for the tone is 9.
325    ZN = 2^BN:ZT = 2^BT
330    B0 = 10:B1 = 11:B2 = 12:B3 = 13:B4 = 14
       REM Binary output for the five lights.
340    QF = 0
       REM Quiet flag 0=noise, 1=quiet.
```

500 - Menu

```
500    HOME: PRINT "**MAIN MENU**":PRINT
510    PRINT "1: CHANGE PARAMETERS": PRINT
520    PRINT "2: RUN SUBJECT": PRINT
530    PRINT "3: CALIBRATE": PRINT
600    GET A$
       REM Get the option from menu.
610    X = VAL (A$)
       REM Give that option a number value.
620    IF X <1 OR X> 3 THEN 600
630    ON X GOTO 1000,2000,3000
```

1000 - Change Parameters

```
1000   GOSUB 17500
1019   PRINT : PRINT "  'M' FOR MAIN MENU"
1050   GET A$: IF A$ = "M" THEN 500
1060   X = VAL (A$): IF X < 1 OR X > 9 THEN 1050
1065   HOME
1070   ON X GOTO 1100,1200,1300,1400,1500,1600,1700,
       1800,1900

1100   PRINT "DURATION OF STIMULUS";DN;"SEC"
1110   PRINT : INPUT "NEW VALUE (0.1 TO 5 SEC)   ";DN
1120   IF DN < .1 OR DN >5 THEN PRINT: PRINT "NOT A
       VALID VALUE": PRINT : PRINT "PLEASE TRY AGAIN!":
       & PAUSE = 2: HOME : PRINT: GOTO 1100
1130   GOTO 1000

1200   PRINT "INTERSTIMULUS INTERVAL      ";DI;"SEC"
1210   PRINT : INPUT "NEW VALUE (0.4 TO 5 SEC)     ";DI
1220   IF DI < .4 OR DI > 5 THEN PRINT: PRINT "NOT A
       VALID VALUE." :PRINT : PRINT "PLEASE TRY AGAIN!":
       & PAUSE = 2: HOME : PRINT GOTO 1210
1230   GOTO 1000
```

```
1300  PRINT "STIMULI PER TRIAL   ";NT
1310  PRINT : INPUT "NEW VALUE (2 TO 5)    ";NT
1320  IF NT <2 OR NT >5 THEN PRINT " PRINT "NOT A VALID
      VALUE." : PRINT : PRINT "PLEASE TRY AGAIN!": &
      PAUSE = 2: HOME : PRINT : GOTO 1310
1330  GOTO 1000

1400  PRINT "HITS TO REVERSE    ";NS
1410  PRINT : INPUT "NEW VALUE (1 TO 5)    ";NS
1420  IF NS <1 OR NS >5 THEN PRINT " PRINT "NOT A VALID
      VALUE." : PRINT : PRINT "PLEASE TRY AGAIN!": &
      PAUSE = 2: HOME : PRINT : GOTO 1410
1430  GOTO 1000

1500  PRINT "REVERSALS AT MIN TO QUIT ";RQ
1510  PRINT : INPUT "NEW VALUE (1 TO 25)       ";RQ
1520  IF RQ <1 OR RQ >25 THEN PRINT " PRINT "NOT A
      VALID VALUE." : PRINT : PRINT "PLEASE TRY
      AGAIN!": & PAUSE = 3: HOME : PRINT : GOTO 1510
1530  GOTO 1000

1600  PRINT "STARTING ATTENUATION     ";AC;"  DB"
1610  PRINT : INPUT "NEW VALUE (0 TO 70 DB)     ";AC
1620  IF AC <0 OR AC >70 THEN PRINT " PRINT "NOT A
      VALID VALUE." : PRINT : PRINT "PLEASE TRY
      AGAIN!": & PAUSE = 2: HOME : PRINT : GOTO 1610
1630  GOTO 1000

1700  PRINT "STARTING STEPSIZE      ";SC;"  DB"
1710  PRINT : INPUT "NEW VALUE (10 TO 40 DB)     ";SC
1720  IF SC <10 OR SC >40 THEN PRINT " PRINT "NOT A
      VALID VALUE." : PRINT : PRINT "PLEASE TRY
      AGAIN!": & PAUSE = 2: HOME : PRINT : GOTO 1710
1730  GOTO 1000

1800  PRINT "MINIMUM STEPSIZE      ";SM;"   DB"
1810  PRINT : INPUT "NEW VALUE (0.5 TO 5 DB)     ";SM
1820  IF SM <0.5 OR SM >5 THEN PRINT " PRINT "NOT A
      VALID VALUE." : PRINT : PRINT "PLEASE TRY
      AGAIN!": & PAUSE = 2: HOME : PRINT : GOTO 1810
1830  GOTO 1000

1900  QF = QF-1: IF QF < 0 THEN QF = 1
1910  GOTO 1000
```

2000 -Run Subject

```
2000  HOME
2010  INPUT "SUBJECT NAME?   ";SN$: PRINT
2020  INPUT "GAP DURATION (MSEC)    ";DG: PRINT
2030  IF DG < 0 OR DG > 188 THEN  PRINT "GAP MUST BE
      BETWEEN 0 TO 188 MSECS": PRINT : GOTO 2020
2032  IF DG / 2 - INT (DG / 2) < > 0 THEN PRINT "EVEN
      NUMBERS": PRINT : GOTO 2020
2035  GOSUB 19000
2040  INPUT "TONE FREQUENCY?   ";F: PRINT
2050  IF F < 125 OR F > 8000 THEN PRINT "SELECT
      FREQUENCY FROM 125 TO 8000 HZ": PRINT : HOME :
      GOTO 2040
      REM Check for frequency mistakes and prompts to
      change manually the frequency sweep range if
      needed.
2060  T$ = "B": IF F > = 3000 THEN T$ = "C"
      REM  The frequency sweep range of the tone is 'C'
      for frequencies above 3000 hz. any frequency less
      than that is at 'B'.
2070  PRINT "HIT 'SPACE BAR' WHEN TONE GENERATOR":
      PRINT "IS SET TO ";T$;", BIPOLAR": GET A$
2075  GOSUB 18000

2080  PRINT : PRINT "MEASURED FREQUENCY = ";FZ
      REM Prints out the measured frequency.
2090  IF ABS ((FZ - F) / F) > .02 THEN PRINT "ERROR
      EXCEEDS 2%": PRINT " HOME : GOTO 2070
```

2100 - Calibration of Tone and Noise

```
2100  REM  Manual calibration for noise alone is 1 volt
      and for tone alone is 300 mv
2105  PRINT : PRINT "REMOVE PHONES FROM SUBJECT": PRINT
      : PRINT "HIT 'SPACE BAR' TO CONTINUE": GET A$

2110  & BOUT,(DV) = 2 ^ BT
      REM  Tone on with 0 db attenuation.
2120  HOME : VTAB 10
2130  PRINT "ADJUST VOLUME CONTROL ON MIXER AMPLIFIER
      SO THAT INPUT TO MANUAL ATTENUATOR" : PRINT "IS
      300 MILLIVOLTS"
2140  PRINT : PRINT "HIT 'SPACE BAR' TO CONTINUE": GET
      A$
2150  HOME
2160  & BOUT, (DV) = 2 ^BN
      REM  Tone is turned off and noise is turned on.
2170  VTAB 8
2180  PRINT "ADJUST NOISE GENERATOR SO THAT INPUT TO
      MANUAL ATTENUATOR IS 1.0 VOLT"
2190  PRINT : PRINT "HIT 'SPACE BAR' TO CONTINUE"
```

```
2192  GET A$
2195  & BOUT,(DV) = 0

2200 - Start Testing
2200  HOME : VTAB 12
2210  INPUT "ENTER SETTING OF MANUAL ATTENUATOR IN
      DB";AM
2220  HOME : GOSUB 17000
2230  PRINT : PRINT "IS THIS CORRECT (Y/N)? ": GET A$:
      IF A$ = "N" THEN 2000
      REM  Repeats the run subject routine.
2240  IF A$ < > "Y" THEN 2230
      REM Makes sure you press "y", otherwise asks the
      question again.
2245  & BOUT,(DV) = 2 ^ 11
2250  HOME : VTAB 10: PRINT "HIT 'READY' LIGHT TO
      BEGIN"
2270  & LOOK FOR BIN, (TV) = WA, (TH) = 2 ^ 11, (XM) =
      65535
      REM Looks for the subjects response to the first
      warning light.
2280  & BOUT, (DV) = 0
      REM Turns first ready light off.
2290  AC = AC - SC

2300  & TIME TO H1,M1,S1
      REM Time when testing begins.

2430  AC = AC + PAD * SC:AC = INT(AC + .5)
2440  NA = NA + 1: IF NA > 50 THEN 2920
2450  Y(NA) = AC

2500  REM Check attenuation.
2510  IF AC > 80 THEN AC = 80
2520  IF AC < 0 THEN AC = 0
2522  Y(NA) = AC
2525  BO = AC * 2: GOSUB 19000
2530  N = 0
      REM Set counter for stimulus number.
2540  & BOUT,(DV) = BO
      REM See remark in 2570.
2570  BO = AC *2: & BOUT,(DV) = BO
      REM BO=number of 1/2 db steps of attenuation.
      Feedback light is turned off as a warning light
      to next stimulus.
2580  GOSUB 11000
      REM Present stimuli.

2610  N = N + 1
      REM Adds the stimulus presentations.
2700  IF H = 0 AND PAD = -1 THEN GOTO 2430
```

```
      REM Increasing attenuation due to a miss and was
      already ascending.
2710  IF H = 0 AND PAD = 1 THEN 2850
      REM Miss on descent, reverse and halve stepsize.
2720  IF H = 1 AND N < NS THEN & PAUSE = 1: GOTO 2540
      REM A hit before maximum stimuli have been
      presented and therefore repeat. Pause is to
      extend feedback.
2730  IF H = 1 AND N = NS AND PAD = 1 THEN GOTO 2430
      REM All hits on descent and therefore continue
      descending.
2740  IF H = 1 AND N = NS AND PAD = -1 THEN GOTO 2850
      REM All hits on ascent and therefore reverse to
      descent and halve stepsize.

2850  SC = SC / 2: IF SC < SM THEN SC = SM
      REM Halve stepsize and check to see if it is at
      minimum.
2880  PAD = PAD * ( - 1)
      REM Reverse direction.
2890  IF SC = SM THEN R = R + 1
      REM Increase reversal counter if minimum stepsize
      has been reached.

2900  X(R) = AC: PRINT AC
      REM Store value of attenuation of r-th reversal.
2905  PRINT "DB": PRINT
2910  IF R < RQ THEN GOTO 2430
2915  & BOUT, (DV) = 0: & TIME TO H2,M2,S2
      REM Time of end of testing (run).
2920  GOSUB 16000
      REM Printing data.
2930  INPUT "'RETURN' FOR MENU";Z$: GOTO 200
3000  GOTO 10000

10000 HOME
10010 VTAB 10
      REM Start ten lines from beginning of page.
10020 PRINT "OPTION NOT YET AVAILABLE, TRY LATER"
10030 & PAUSE = 1
10040 GOTO 500

11000 PRINT BO
11010 OT = INT (1 + NT * RND (1))
      REM Generates a random number from 1 to NT.
11015 PRINT OT
11020 FOR X = 1 TO NT
11025 BB = BN: IF QF = 1 THEN BB = 10 + X
      REM When tone is tested in silence then the
      lights are turned on in each trial to cue subject
      for the trial.
```

```
11030 IF X = OT THEN GOSUB 13000: GOTO 11040
11035 GOSUB 12000
11040 NEXT X
11045 REM GET SUBJECT'S RESPONSE
11050 & LOOK FOR BIN,(TV) = BR,(TH) = 2 ^ 11,(XM) =
      65535
11060 SR = BR / 2 ^ 10
11070 SR = LOG (SR) / LOG (2)
      REM Done so that subject's response (SR) will
      have a value of 1,2,3,4, or 5.
11080 H = 0: IF SR = OT THEN H = 1
      REM Gives a value of 1 for a hit, otherwise
      remains 0.
11090 & BOUT,(DV) = BO + 2 ^ (10 + OT)
      REM The light corresponding to the right answer
      (i.e. interval containing tone) is turned on.
11120 RETURN

12000 REM SUB-ROUTINE FOR GAP WITHOUT TONE
12010 & PAUSE = DI
      REM DI is interstimulus duration.
12015 PRINT X,
12020 & BOUT,(DV) = BO + 2 ^ BB
      REM Turns noise on with current attenuation
      level.
12030 & PAUSE = DN / 4
      REM Noise is on for DN/4 seconds.
12040 & BOUT,(DV) = BO + 2 ^ BB
12050 & PAUSE = DN / 4
      REM Noise is on for another DN/4 seconds but
      without tone to make sure that this and
      subroutine 13000 will be equal in duration.
12060 & @BOUT,(AV) = GS,(RT) = 2
      REM Array variable of silent gap, noise is turned
      off at rate = RT, each 1 or 0 of array
      corresponds to 2 msec.
12070 & PAUSE = DN / 4
      REM Noise continues for another DN/4 seconds.
12080 & BOUT,(DV) = BO + 2 ^ BB
      REM In subroutine 13000 here is where the tone is
      turned off but noise continues.
12090 & PAUSE = DN / 4
      REM Noise is on for DN/4 seconds.

12100 & BOUT,(DV) = BO
      REM Noise is turned off, but current attenuation
      level of tone remains see rem 13100.
12110 RETURN

13000 REM SUBROUTINE FOR GAP WITH TONE
13010 & PAUSE = DI
```

```
      REM Interstimulus duration.
13015 PRINT X,
13020 & BOUT,(DV) = BO + 2 ^ BB
      REM NOise is turned on with the current
      attenuation level.
13030 & PAUSE = DN / 4
      REM The noise is on by itself for DN/4 seconds.
13040 & BOUT,(DV) = BO + 2 ^ BB + 2 ^ BT
      REM Both noise and tone are turned on with the
      current attenuation level.
13050 & PAUSE = DN / 4
      REM the noise and tone on for dn/4 seconds.
13060 & @BOUT,(AV) = GT,(RT) = 2
      REM Array variable for silent gap with tone.
13070 & PAUSE = DN / 4
      REM Tone and noise are one for dn/4 seconds.
13080 & BOUT,(DV) = BO + 2 ^ BB
      REM Tone is turned off, only noise is on with
      current attenuation level.
13090 & PAUSE = DN / 4
      REM Noise is on for DN/4 seconds.
13100 & BOUT,(DV) = BO
      REM Noise is turned off, but the current
      attenuation value of the tone remains. The next
      time the tone is turned on it will have that
      value.
13110 RETURN

14000 REM SET BINARY OUTPUT FOR CURRENT VALUE OF
      ATTENUATION
14010 REM Assumes that binary outputs 0 thru 7 control
      attenuation in 0.5 db steps.
14020 BO = AC * 2
      REM BO=number of 1/2 db steps of attenuation.
14030 & BOUT,(DV) = BO
14040 RETURN

16000 REM PRINT REVERSAL DATA
16010 PR#1: GOSUB 17000: PRINT: PRINT : GOSUB 17500
16015 GOTO 16100
16020 PRINT : PRINT "REVERSALS:"
16030 S = 0
16050 FOR C = 2 TO RQ
16060 S = S + X(C)
16070 PRINT X(C)
16080 NEXT C
16090 MN = INT (S / (RQ - 1) * 10 + .5) / 10: PRINT :
      PRINT "MEAN: ";MN;" DB"
16095 GOTO 16200

16100 PRINT : PRINT "ALL DATA"
```

```
16105 PR#0: INPUT "HIT 'ENTER' WHEN PAPER IS SET IN
      PRINTER";Z$: PR #1
16110 FOR C = 1 TO NA
16115 IF C < 10 THEN PRINT " ";
16120 PRINT C;":"; SPC( Y(C) / 2);Y(C):
      REM Prints the graph.
16130 NEXT C
16140 GOTO 16020
16200 PR #0: RETURN

17000 REM SUBROUTINE TO GET PRINTED INFORMATION
17005 & DAY TO YR,MT,DY,DA
17007 PRINT "DATE OF TEST  ";MT;"/";DY;"/";YR
17010 PRINT : PRINT "SUBJECT NAME ";SN$
17020 PRINT : PRINT "GAP DURATION ";DG;" MSEC"
17030 PRINT : PRINT "FREQUENCY";F;" (";FZ;") HZ"
17040 PRINT : PRINT "MANUAL ATTEN ";AM;" DB"
17050 PRINT : PRINT "START TIME";H1;":";M1;":";S1
17060 PRINT : PRINT "END TIME";H2;":";M2;":";S2
17070 TT = (S2 + 60 * M2 + 60 * 60 * H2) - (S1 + 60 *
      M1 + 60 * 60 * H1)
      REM TT = Testing time in seconds.
17080 M = INT (TT / 60):SL = TT - 60 * M
      REM Converting the testing time in seconds back
      to minutes.
17090 PRINT : PRINT "TIME TAKEN ";M;" MIN AND ";SL;"
      MSEC
17100 RETURN
```

<u>Print of Select parameters</u>

```
17500 HOME
17510 PRINT "1: DURATION OF STIMULUS        ";DN;"
      SEC": PRINT
17520 PRINT "2: INTERSTIMULUS INTERVAL       ";DI;"
      SEC": PRINT
17530 PRINT "3: STIMULI PER TRIAL     ";NT: PRINT
17540 PRINT "4: HITS TO REVERSE      ";NS: PRINT
17550 PRINT "5: REVERSALS AT MIN TO QUIT      ";RQ:
      PRINT
17560 PRINT "6: STARTING ATTENUATION      ";AS;"  DB":
      PRINT
17570 PRINT "7: STARTING STEPSIZE       ";SS;"  DB":
      PRINT
17580 PRINT "8: MINIMUM STEPSIZE       ";SM;"  DB":
      PRINT
17590 PRINT "9: TONE IN  ";
17595 IF QF = 0 THEN PRINT "NOISE"
17597 IF QF = 1 THEN PRINT "QUIET"
17600 RETURN
```

<u>Calibrate frequency</u>

```
18005 V1 = 1700:V2 = 2300
18010 & AOUT,(C#) = 0,(DV) = V1
      REM AOUT 0 connects to analog input of tone
      generator from computer and gives the lower of
      two frequencies a number - v1 in this case.
18020 & FINL,(TV) = FL,(C#) = 0
      REM FL is the lower frequency of two frequencies
18030 & AOUT,(C#) = 0,(DV) = V2
      REM The higher frequency is assigned a computer
      #V2
18040 & FINL,(TV) = FH,(C#) = 0
      REM FH is the higher number of two frequencies
18050 V = V1 + (V2 - V1) * (F - FL) / (FH - FL)
      REM The number/voltage of the desired frequency.
18060 & AOUT,(C#) = 0,(DV) = V
      REM Logic out of tone generator (via -12 to +5).
      to counter in #0. The calculated voltage/number
      for the desired frequency is entered.
18070 & FINL,(C#) = 0,(TV) = FZ
      REM FZ is the frequency of the signal with a
      voltage of v that is entered in counter in 0.
18080 RETURN

19000 TZ = ZN + ZT + BO: SZ = ZN + BO
19005 FOR X = 0 TO 100
19010 GT(X) = TZ
      REM Gap array with tone and noise.
19020 GS(X) = SZ
      REM A gap array with noise only.
19030 NEXT X
19040 IF DG = 0 THEN GOTO 19100
19050 REM Z = INT(100 - DG / 2) / 2
19055 Z = INT (50 - DG / 4)
19060 FOR X = Z TO Z + (DG / 2 - 1)
19070 GT(X) = ZT + BO
19080 GS(X) = BO
19090 NEXT X
19100 RETURN
```

APPENDIX H

INSTRUCTIONS

The following are the written instructions given to the subjects prior to testing:

"YOU WILL BE HEARING TWO INTERVALS OF NOISE WHICH WILL ALSO BE INDICATED BY FLASHING LIGHTS ON THE RESPONSE BOX PLACED IN FRONT OF YOU. ONLY ONE OF THE INTERVALS OF NOISE CONTAINS A TONE. YOU ARE TO DETERMINE WHICH OF THE TWO INTERVALS CONTAINS THE TONE: INTERVAL 1 OR INTERVAL 2. IF YOU THINK YOU HEARD THE TONE IN THE FIRST INTERVAL, PLEASE PRESS KEY #1 ON THE RESPONSE BOX. IF YOU THINK YOU HEARD THE TONE IN THE SECOND INTERVAL, PLEASE PRESS KEY #2 ON THE RESPONSE BOX. AFTER YOU HAVE RESPONDED, YOU WILL BE PROVIDED WITH FEEDBACK BY THE LIGHTS ON THE RESPONSE BOX. THAT IS, IF THE TONE WAS IN THE FIRST INTERVAL, THE FIRST LIGHT WILL BE TURNED ON, WHEREAS, IF THE TONE WAS IN THE SECOND INTERVAL, THE SECOND LIGHT WILL BE TURNED ON. IMMEDIATELY AFTER THE PROVISION OF FEEDBACK, YOU WILL HEAR THE NEXT PAIR OF NOISE INTERVALS OF WHICH ONLY ONE CONTAINS A TONE. AGAIN, YOU WILL HAVE TO KEY IN YOUR RESPONSE. NOTE THAT THE TEST WILL NOT PROCEED UNLESS YOU RESPOND. PLEASE RESPOND ON EVERY TRIAL EVEN IF THE TONE IS SO FAINT THAT YOU HAVE TO GUESS. IF YOU HAVE ANY QUESTIONS REGARDING THE INSTRUCTIONS OF THE

TEST AND/OR THE TEST ITSELF, PLEASE DO NOT HESITATE TO ASK ME."

APPENDIX I

RAW DATA

Appendix I.1

Subject LK: Thresholds of Tones (in dB Attenuation) for Individual Runs, Means, Standard Deviation (SD), and Standard Error (SE) at 9 Gap Values and in Quiet. Last Table: Mean Threshold re: Mean Threshold in Noise.

0.5 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	24.20	22.30	27.70	26.20	37.70	37.00	47.00	50.00	50.00	58.30
	25.00	21.70	24.70	27.60	34.30	39.70	48.70	50.00	47.30	57.30
	27.70	26.30	27.70	29.00	35.30	38.00	51.70	47.30	51.70	57.00
	24.30	22.70	23.34	29.30	37.70	38.70	46.00	47.00	51.00	61.00
	26.70	22.00	25.00	27.00			46.70			
	27.20						47.20			
	24.90									
	25.30									
MEAN	25.66	23.00	25.69	27.82	36.25	38.35	47.88	48.58	50.00	58.40
SD	1.35	1.88	1.94	1.32	1.72	1.14	2.07	1.65	1.93	1.82
SE	0.48	0.84	0.87	0.59	0.86	0.57	0.85	0.82	0.97	0.91

1.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	23.70	24.00	33.30	39.20	45.70	52.70	55.70	59.00	60.00	74.30
	25.30	25.00	33.00	41.00	48.00	55.00	57.30	61.00	59.70	70.70
	24.30	25.00	35.70	45.00	45.70	53.00	58.30	62.00	62.00	68.30
	23.20	26.70	33.30	39.70	46.70	50.30	56.30	59.30	62.00	66.70
	24.40			41.00	50.70					69.30
	25.70			42.40	50.70					69.30
				42.70						71.00
MEAN	24.43	25.18	33.83	41.57	47.92	52.75	56.90	60.33	60.93	69.94
SD	0.94	1.12	1.26	1.98	2.32	1.93	1.14	1.42	1.25	2.41
SE	0.38	0.56	0.63	0.75	0.95	0.96	0.57	0.71	0.62	0.91

2.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	18.80	20.30	35.70	40.60	47.30	47.30	53.70	51.30	58.70	60.00
	19.00	21.70	33.00	44.30	47.00	46.70	53.70	52.00	56.00	60.00
	20.40	20.00	37.30	42.30	43.30	47.70	52.00	50.70	53.30	61.30
	19.70	18.30	34.70	41.70	44.00	47.00	49.70	53.00	55.00	60.30
	21.70	23.30	35.30	41.70	43.30				55.30	
	20.70	19.00	34.00	41.00	44.70					
	19.10			43.00						
				40.00						
MEAN	19.92	20.43	35.00	41.83	44.93	47.18	52.28	51.75	55.66	60.40
SD	1.07	1.82	1.48	1.38	1.80	0.43	1.89	0.99	1.97	0.62
SE	0.40	0.74	0.60	0.49	0.73	0.21	0.95	0.49	0.88	0.31

4.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	17.40	23.00	35.70	40.60	38.30	45.70	46.30	51.00	51.70	59.70
	17.80	23.70	38.00	36.00	41.00	43.30	47.70	53.30	53.30	59.30
	15.00	23.00	38.30	36.00	42.70	41.30	46.30	55.00	52.30	61.30
	15.70	19.00	36.00	37.30	44.00	45.00	45.30	51.10	52.00	63.30
	19.00	19.00		42.00	40.00					55.30
	15.30	22.00		35.30	39.30					61.00
	15.70			37.30						55.70
	14.70									57.30
	16.00									61.30
MEAN	16.29	21.62	37.00	37.79	40.88	43.83	46.40	52.60	52.33	59.37
SD	1.45	2.10	1.34	2.54	2.14	1.96	0.99	1.92	0.69	2.74
SE	0.48	0.86	0.67	0.96	0.88	0.98	0.49	0.96	0.35	0.91

MEAN THRESHOLD re: MEAN THRESHOLD in NOISE (Gap = 0 msec)

	ACTUAL GAP (in msec)								
	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
0.5 KHz	-2.66	0.03	2.16	10.59	12.69	22.22	22.92	24.34	32.74
1.0 KHz	0.75	9.40	17.14	23.49	28.32	32.47	35.90	36.50	45.51
2.0 KHz	0.51	15.08	21.91	25.01	27.26	32.36	31.83	35.74	40.48
4.0 KHz	5.33	20.71	21.50	24.59	27.54	30.11	36.31	36.04	43.08

Appendix I.2

Subject EL: Thresholds of Tones (in dB Attenuation) for Individual Runs, Means, Standard Deviation (SD), and Standard Error (SE) at 9 Gap Values and in Quiet. Last Table: Mean Threshold re: Mean Threshold in Noise.

0.5 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	23.70	26.20	22.70	31.00	29.70	36.00	42.70	45.30	39.70	48.70
	24.70	24.30	24.30	31.30	30.30	42.00	40.70	42.20	46.00	46.70
	22.70	24.30	23.30	31.00	32.70	39.70	39.30	41.30	43.00	51.70
	21.30	22.30	24.30	27.30	29.70	38.30	40.00	42.00	38.70	47.70
	21.70			29.70		37.60			40.30	49.00
	22.70					40.00			42.00	
									45.00	
MEAN	22.80	24.28	23.65	30.06	30.60	38.93	40.68	42.70	42.10	48.76
SD	1.26	1.59	0.79	1.66	1.43	2.09	1.47	1.78	2.74	1.88
SE	0.51	0.80	0.39	0.74	0.71	0.85	0.73	0.89	1.04	0.84

1.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	20.70	18.40	34.00	42.30	41.30	41.80	46.90	49.00	55.70	64.70
	22.30	22.00	32.30	41.30	40.00	43.30	49.00	53.00	55.30	59.00
	20.70	21.30	30.30	38.00	45.30	43.30	49.70	48.70	54.70	63.00
	25.70	21.30	30.30	37.00	43.70	47.00	47.30	48.70	55.00	63.10
	24.30			38.00	41.00	44.00		50.70		62.00
	23.30			38.30						
	22.30									
	23.30									
MEAN	22.83	20.75	31.73	39.15	42.26	43.88	48.23	50.02	55.18	62.36
SD	1.71	1.60	1.79	2.12	2.18	1.92	1.34	1.86	0.43	2.11
SE	0.34	0.80	0.89	0.87	0.97	0.86	0.67	0.83	0.21	0.94

2.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	21.00	22.70	34.70	35.00	41.30	40.70	50.00	50.30	52.00	63.00
	23.70	21.70	34.00	35.00	40.70	42.00	47.00	49.00	56.00	59.00
	20.70	26.70	32.70	38.70	45.00	43.00	46.70	52.00	53.30	62.70
	21.70	27.70	28.70	37.30	41.70	43.00	46.30	51.30	54.30	62.00
	21.00	25.00	32.00	37.00	40.00					
	22.70	24.30	32.00	37.30						
MEAN	21.78	24.68	32.35	36.72	41.74	42.18	47.50	50.65	53.90	61.68
SD	1.18	2.29	2.09	1.46	1.93	1.09	1.69	1.30	1.69	1.83
SE	0.48	0.94	0.86	0.59	0.86	0.55	0.85	0.65	0.84	0.92

4.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	13.70	16.20	34.00	38.00	38.70	41.60	46.70	54.00	58.30	66.00
	15.70	14.30	33.70	41.00	40.70	47.70	47.00	52.00	59.00	65.00
	16.30	19.00	36.30	36.30	40.00	46.70	50.30	53.70	57.70	62.00
	16.00	18.00	30.70	36.30	39.00	41.30	47.00	48.30	58.00	62.70
	14.30	16.30	34.20	40.70		41.60	47.00	51.70		
	16.60	17.50		37.70		42.30		52.00		
						43.00				
						46.70				
MEAN	15.43	16.88	33.78	38.33	39.60	43.86	47.60	51.95	58.25	63.93
SD	1.17	1.65	2.00	2.07	0.92	2.69	1.51	2.03	0.56	1.89
SE	0.48	0.67	0.90	0.85	0.46	0.95	0.68	0.83	0.28	0.94

MEAN THRESHOLD re: MEAN THRESHOLD in NOISE (Gap = 0 msec)

	ACTUAL GAP (in msec)								
	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
0.5 KHz	1.48	0.85	7.26	7.80	16.13	17.88	19.90	19.30	25.96
1.0 KHz	-2.08	8.90	16.32	19.43	21.05	25.40	27.19	32.35	39.53
2.0 KHz	2.90	10.57	14.94	19.96	20.40	25.72	28.87	32.12	39.90
4.0 KHz	1.45	18.35	22.90	24.17	28.43	32.17	36.52	42.82	48.50

Appendix I.3

Subject MB: Thresholds of Tones (in dB Attenuation) for Individual Runs, Means, Standard Deviation (SD), and Standard Error (SE) at 9 Gap Values and in Quiet. Last Table: Mean Threshold re: Mean Threshold in Noise.

0.5 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	22.00	26.00	26.00	30.20	34.00	38.70	46.70	51.30	52.00	61.00
	22.70	24.30	30.70	30.70	31.30	38.00	47.30	49.00	52.00	63.30
	22.30	25.00	27.70	30.00	31.70	38.00	47.00	50.30	50.30	61.30
	25.00	26.00	26.70	32.70	35.70	35.30	46.70	47.30	52.30	63.00
	25.70		27.75	31.70	33.20					
	23.70									
	23.00									
	22.00									
MEAN	23.30	25.33	27.78	31.06	33.18	37.50	46.93	49.48	51.65	62.15
SD	1.40	0.83	1.79	1.13	1.79	1.50	0.29	1.73	0.91	1.17
SE	0.49	0.42	0.80	0.50	0.80	0.75	0.14	0.86	0.46	0.58

1.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	21.80	23.00	30.30	41.30	41.00	46.00	50.30	56.00	57.30	72.00
	21.00	21.30	28.70	36.00	41.70	44.30	49.30	54.70	57.30	66.60
	20.70	20.00	28.70	35.00	39.30	42.00	48.70	54.30	55.30	66.00
	22.00	21.00	30.30	39.00	38.70	43.70	50.00	55.30	57.00	70.10
				36.00						68.30
				35.00						71.20
				36.90						
MEAN	21.38	21.33	29.50	37.05	40.18	44.00	49.58	55.08	56.73	69.03
SD	0.62	1.25	0.92	2.33	1.41	1.65	0.72	0.74	0.96	2.46
SE	0.31	0.62	0.46	0.95	0.70	0.83	0.36	0.37	0.48	1.00

2.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	17.30	20.70	33.00	35.60	44.70	44.70	46.00	54.30	51.00	63.70
	21.00	20.30	33.30	35.70	41.00	42.70	46.30	50.30	53.30	68.30
	17.70	17.70	30.70	34.70	40.30	43.30	47.30	52.70	55.00	64.00
	19.70	19.70	32.70	38.00	41.00	40.70	48.70	51.70	53.70	65.20
	18.00									65.50
	17.00									
	18.30									
	20.00									
MEAN	18.63	19.60	32.43	36.00	41.75	42.85	47.08	52.25	53.25	65.30
SD	1.44	1.33	1.18	1.41	1.99	1.66	1.22	1.68	1.67	1.82
SE	0.51	0.67	0.59	0.70	1.00	0.83	0.61	0.84	0.83	0.82

4.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	16.00	15.70	24.70	30.40	34.00	40.70	44.70	47.30	48.70	57.00
	14.00	15.70	28.70	30.70	34.00	41.00	46.00	47.00	50.30	60.70
	14.70	19.30	27.70	29.00	33.70	42.00	40.00	50.70	44.00	58.00
	17.70	19.00	26.70	32.70	36.00	41.70	42.30	47.00	48.00	58.30
	15.30						42.70		46.60	
	15.00								46.50	
	17.70									
	16.00									
MEAN	15.80	17.43	26.95	30.70	34.43	41.35	43.14	48.00	47.35	58.50
SD	1.34	2.00	1.71	1.53	1.06	0.60	2.31	1.81	2.17	1.57
SE	0.48	1.00	0.85	0.76	0.53	0.30	1.03	0.90	0.88	0.78

MEAN THRESHOLD re: MEAN THRESHOLD in NOISE (Gap = 0 msec)

	ACTUAL GAP (in msec)								
	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
0.5 KHz	2.03	4.48	7.76	9.88	14.20	23.63	26.18	28.35	38.85
1.0 KHz	-0.05	8.12	15.67	18.80	22.62	28.20	33.70	35.35	47.65
2.0 KHz	0.97	13.80	17.37	23.12	24.22	28.45	33.62	34.62	46.67
4.0 KHz	1.63	11.15	14.90	18.63	25.55	27.34	32.20	31.55	42.70

Appendix I.4

Subject OE: Thresholds of Tones (in dB Attenuation) for Individual Runs, Means, Standard Deviation (SD), and Standard Error (SE) at 9 Gap Values and in Quiet. Last Table: Mean Threshold re: Mean Threshold in Noise.

0.5 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	21.00	21.30	27.00	27.30	30.40	32.70	37.30	38.70	44.30	48.70
	20.30	25.00	30.00	31.70	31.70	34.70	37.00	38.00	41.30	50.70
	19.30	20.70	26.70	28.00	35.00	32.30	36.00	40.00	46.00	49.00
	21.00	22.00	25.70	30.70	33.30	35.00	35.30	41.30	43.00	51.00
		24.00		28.30						
MEAN	20.40	22.60	27.35	29.20	32.60	33.68	36.40	39.50	43.65	49.85
SD	0.80	1.83	1.85	1.89	1.99	1.37	0.92	1.46	1.99	1.17
SE	0.40	0.82	0.93	0.85	1.00	0.69	0.46	0.73	1.00	0.58

1.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	20.00	18.30	34.70	44.00	43.80	44.70	51.70	52.70	57.70	65.70
	18.00	20.00	36.30	42.30	41.70	47.00	50.70	56.30	61.70	68.00
	20.70	21.00	29.70	42.00	43.00	46.70	50.70	52.00	57.00	67.70
	17.70	18.70	30.70	41.00	44.00	47.30	53.30	54.00	58.70	68.30
	20.30		35.00						62.30	
	21.30		35.70						58.00	
	17.00		31.70							
	18.70									
	19.00									
MEAN	19.19	19.50	33.40	42.33	43.13	46.43	51.60	53.75	59.23	67.13
SD	1.47	1.24	2.64	1.25	1.04	1.18	1.23	1.89	2.22	1.18
SE	0.49	0.62	1.00	0.62	0.52	0.59	0.61	0.95	0.91	0.59

2.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	13.30	22.30	36.30	37.10	40.30	45.00	43.30	48.70	47.00	63.00
	14.00	20.00	36.70	37.80	39.00	44.00	45.30	44.70	51.30	58.70
	15.30	21.70	34.00	37.00	41.00	42.30	46.30	45.30	45.70	60.70
	16.00	19.30	35.30	38.30	38.30	46.00	42.30	46.70	44.30	59.00
	17.00		34.30						49.00	
	16.30								47.00	
	17.70									
	15.30									
	17.00									
MEAN	15.77	20.83	35.32	37.55	39.65	44.33	44.30	46.35	47.38	60.35
SD	1.45	1.41	1.19	0.61	1.22	1.58	1.83	1.78	2.47	1.97
SE	0.48	0.70	0.53	0.31	0.61	0.79	0.91	0.89	1.01	0.99

4.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	15.00	21.70	41.00	41.30	42.20	50.70	52.30	56.70	53.00	69.70
	15.30	24.00	38.00	41.70	39.30	47.30	55.30	54.70	53.30	69.30
	14.00	18.70	38.30	40.00	40.70	53.30	52.00	55.70	58.00	69.00
	15.00	25.00	38.30	42.20	43.70	51.00	54.00	55.30	56.70	72.70
		23.30				51.00			55.00	
		22.70								
MEAN	14.83	22.57	38.90	41.30	41.48	50.66	53.40	55.60	55.20	70.18
SD	0.57	2.20	1.41	0.94	1.90	2.15	1.54	0.84	2.16	1.71
SE	0.28	0.90	0.70	0.47	0.95	0.96	0.77	0.42	0.96	0.85

MEAN THRESHOLD re: MEAN THRESHOLD in NOISE (Gap = 0 msec)

	ACTUAL GAP (in msec)								
	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
0.5 KHz	2.20	6.95	8.80	12.20	13.28	16.00	19.10	23.25	29.45
1.0 KHz	0.31	14.21	23.14	23.94	27.24	32.41	34.56	40.04	48.24
2.0 KHz	5.06	19.55	21.78	23.88	28.56	28.53	30.58	31.61	44.58
4.0 KHz	7.74	24.07	26.47	26.65	35.83	38.57	40.77	40.37	55.35

Appendix I.5

Subject LH: Thresholds of Tones (in dB Attenuation) for Individual Runs, Means, Standard Deviation (SD), and Standard Error (SE) at 9 Gap Values and in Quiet. Last Table: Mean Threshold re: Mean Threshold in Noise.

0.5 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	25.00	24.20	28.70	25.00	26.00	29.70	36.90	35.00	46.00	56.70
	20.70	24.20	26.00	28.70	27.10	31.70	35.30	34.70	48.30	59.00
	22.30	25.60	23.10	25.80	27.70	31.10	37.60	40.00	43.30	53.30
	19.70	23.10	24.00	26.00	33.00	31.30	35.30	37.70	46.70	57.30
	24.00	23.00	25.00		29.70	29.00		37.30		56.70
	22.70				28.70					
	21.30				27.00					
	21.70				27.30					
	25.70									
	25.00									
	21.30									
	25.00									
	22.00									
	22.30									
MEAN	22.76	24.02	25.36	26.38	28.31	30.56	36.28	36.94	46.08	56.60
SD	1.87	1.05	2.16	1.61	2.20	1.15	1.16	2.17	2.09	2.07
SE	0.50	0.47	0.97	0.80	0.78	0.52	0.58	0.97	1.04	0.93

1.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	21.70	25.00	28.00	32.70	40.70	40.30	46.40	51.30	48.00	61.30
	22.30	27.30	29.00	35.00	35.30	38.90	50.90	48.30	50.00	59.70
	21.70	26.00	28.40	33.30	32.30	40.90	43.30	47.30	50.70	59.00
	20.00	26.70	30.70	36.70	38.40	37.00	50.70	47.70	48.70	61.30
	23.70				33.70		41.70	53.30		60.70
	22.00				41.70		40.30	50.70		59.00
					33.70		49.00			58.70
					37.00		45.30			59.70
					36.30		43.00			
					39.00		43.70			
							44.00			
							42.70			
MEAN	21.90	26.25	29.03	34.43	36.81	39.28	45.08	49.77	49.35	59.93
SD	1.19	0.99	1.19	1.80	3.14	1.73	3.48	2.38	1.22	1.05
SE	0.49	0.49	0.59	0.90	0.99	0.87	1.00	0.97	0.61	0.37

2.0 KHz	ACTUAL GAP (in msec)									
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	22.30	19.60	27.70	35.70	37.60	38.70	48.90	54.70	53.00	71.70
	22.70	21.80	29.30	34.30	39.10	45.70	47.10	58.00	56.70	71.30
	19.70	19.00	30.00	35.60	35.00	40.00	49.30	53.30	55.00	73.70
	22.70	18.00	29.00	37.30	37.00	46.00	45.30	54.00	59.00	73.30
	25.00					43.00		54.00	55.00	
	20.00					40.30			59.30	
	18.30					45.30				
	23.00					40.00				
	18.70					40.70				
	18.70									
	21.70									
	20.00									
	22.70									
	18.70									
	22.00									
	23.30									
	21.60									
MEAN	21.24	19.60	29.00	35.73	37.18	42.19	47.65	54.80	56.33	72.50
SD	1.99	1.61	0.96	1.23	1.70	2.84	1.84	1.86	2.48	1.18
SE	0.48	0.80	0.48	0.61	0.85	0.95	0.92	0.83	1.01	0.59

4.0 KHz	ACTUAL GAP (in msec)								
	0	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
	14.00	19.10	26.30	28.30	30.30	32.00	40.70	41.30	38.00	51.70
	12.00	18.00	24.70	30.00	30.20	33.30	44.70	37.00	43.00	54.00
	11.70	16.00	24.70	28.80	31.30	30.00	40.90	39.30	37.70	50.30
	14.70	16.70	22.70	32.70	32.00	32.00	43.00	35.30	35.30	54.00
	15.30							44.00	39.70	48.30
	12.30							31.30	40.00	51.70
	12.70							36.30	39.30	
	12.00							33.70		
								39.70		
								37.00		
								40.00		
								36.30		
								42.70		
MEAN	13.09	17.45	24.60	29.95	30.95	31.83	42.33	37.99	39.00	52.00
SD	1.38	1.38	1.47	1.97	0.86	1.36	1.89	3.61	2.38	2.19
SE	0.49	0.69	0.74	0.98	0.43	0.68	0.95	1.00	0.90	0.83

MEAN THRESHOLD re: MEAN THRESHOLD in NOISE (Gap = 0 msec)

	ACTUAL GAP (in msec)								
	6.5	17.2	28.0	39.0	49.5	71.0	92.5	114.0	QUIET
0.5 KHz	1.26	2.60	3.62	5.55	7.80	13.52	14.18	23.32	33.84
1.0 KHz	4.35	7.13	12.53	14.91	17.38	23.18	27.87	27.45	38.03
2.0 KHz	-1.64	7.76	14.49	15.94	20.95	26.41	33.56	35.09	51.26
4.0 KHz	4.36	11.51	16.86	17.86	18.74	29.24	24.90	25.91	38.91

Appendix J

The Linear Decay of Energy in the Cochlea on a Log Scale

Consider the energy in the cochlea decays exponentially

$$E = E_o * e^{-\left(\frac{t}{T}\right)} \qquad (J.1)$$

where E = energy in cochlea at time t
E_o = energy in the cochlea at time=0
t = time
T = Time constant

To transform energy to a log scale, by definition

$$L_{dB} = 10\ Log_{10}\left[\frac{E}{Ref}\right] \qquad (J.2)$$

where L = Energy in dB
E = Energy in the cochlea.
Ref = Some reference.

By substituting equation J.1 in equation J.2, then

$$L_{dB} = 10\ Log_{10}\left[\frac{E_o * e^{-\left(\frac{t}{T}\right)}}{Ref}\right] \qquad (J.3)$$

Thus:

$$L_{dB} = 10\ Log_{10}\left(\frac{E_o}{Ref}\right) + 10\ Log_{10}\left[e^{-\left(\frac{t}{T}\right)}\right] \qquad (J.4)$$

Because the variables E_0 and Ref are constants, $10\ Log_{10}\left(\frac{E_o}{Ref}\right)$ can be substituted by the constant b. Also, by transforming equation J.4 to natural logarithms. Then

$$L_{dB} = b + 10\ Log_e\left[e^{-\left(\frac{t}{T}\right)}\right] * 10\ Log_{10}(e) \qquad (J.5)$$

Simplification of Equation J.5 will result with the following equation

$$L_{dB} = b - \left[\frac{10\ \mathrm{Log}_{10}(e)}{T}\right] * t \qquad \text{(J.6)}$$

Because 10 $\mathrm{Log}_{10}(e) = 4.343$, then Equation J.6 can be written as

$$L_{dB} = b - \left[\frac{4.343}{T}\right] * t \qquad \text{(J.7)}$$

where $\left[\frac{4.343}{T}\right]$ = the gradient of the linear function J.7.

APPENDIX K

MEASURED AND PREDICTED DATA

FOR EACH SUBJECT

Table K.1

Total Sum of Squares and Standard Error of Prediction between Measured and Predicted Data (from Equation 5.10) for each subject

Subject	Total sum of squares	$\pm$ Standard Error of prediction
LK	58.51	1.37
EL	48.19	1.25
MB	44.81	1.20
OE	55.74	1.34
LH	75.31	1.56

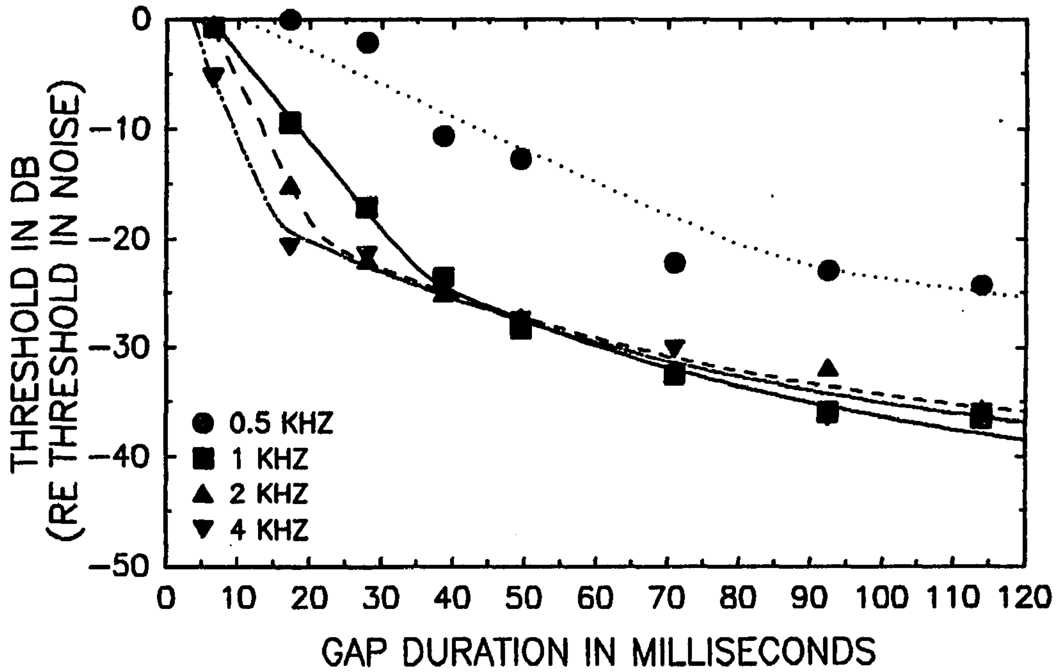

FIGURE K.1: RAW DATA AND BEST FITTING CURVES FOR SUBJECT LK

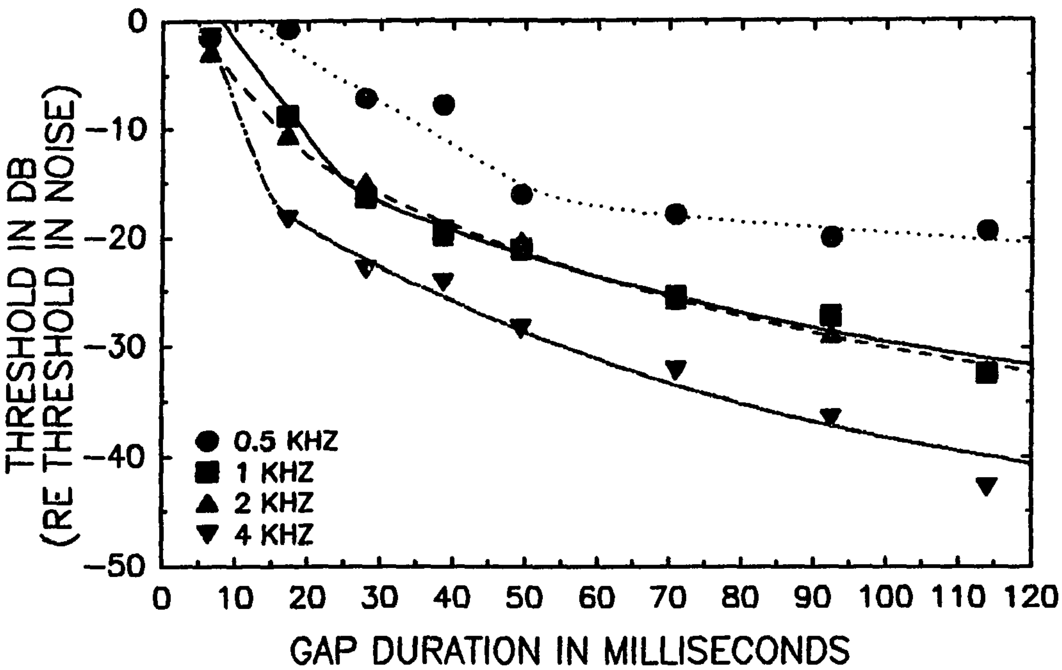

FIGURE K.2: RAW DATA AND BEST FITTING CURVES FOR SUBJECT EL

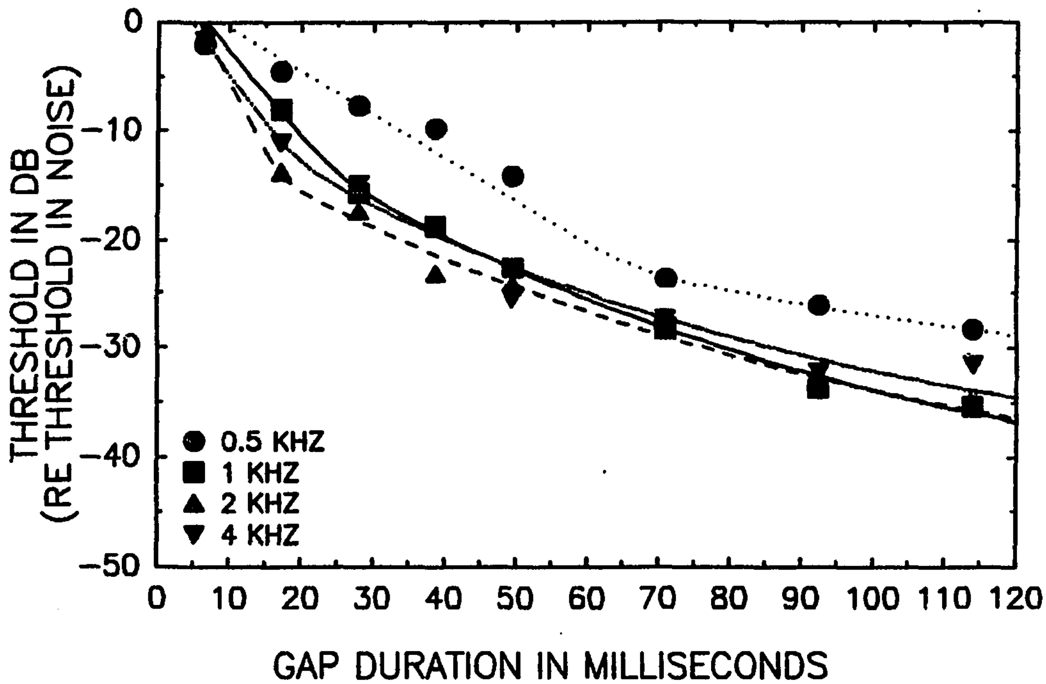

FIGURE K.3: RAW DATA AND BEST FITTING CURVES FOR SUBJECT MB

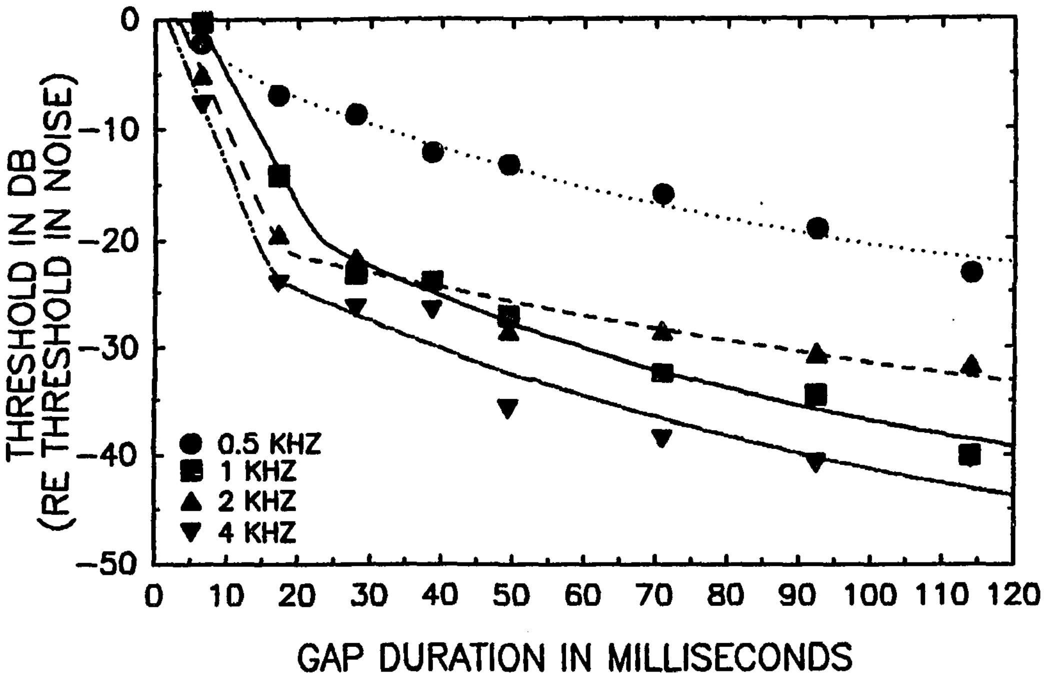

FIGURE K.4: RAW DATA AND BEST FITTING CURVES FOR SUBJECT OE

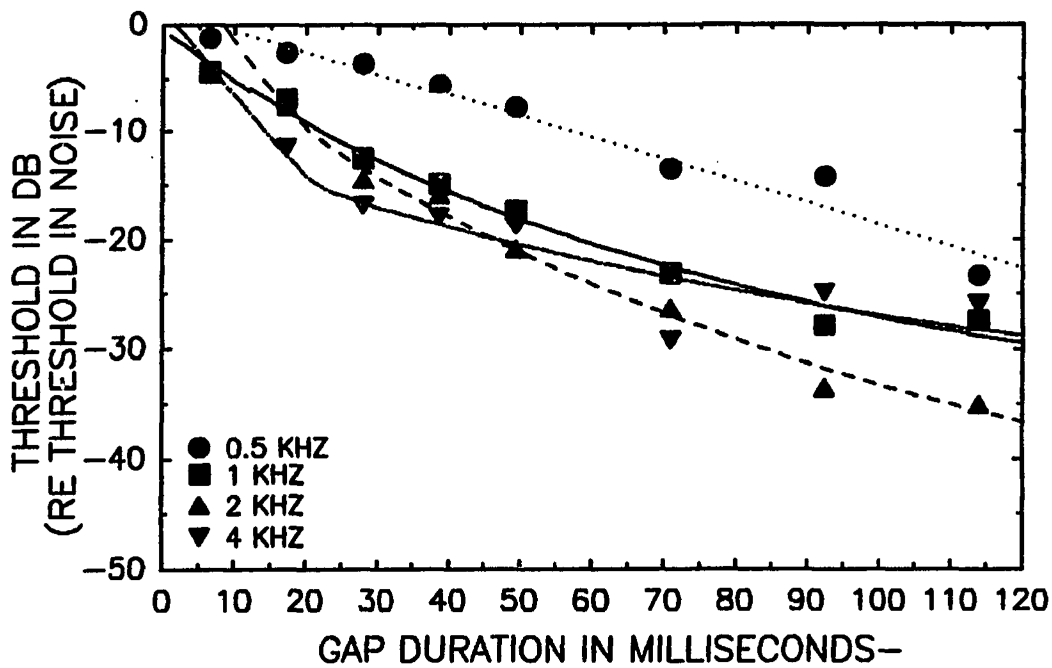

FIGURE K.5: RAW DATA AND BEST FITTING CURVES FOR SUBJECT LH

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