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## Shape discrimination of spectra with various bandwidths

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### 1 Introduction

In what we call 'profile' experiments, the listener's task is to discriminate between two sounds that differ in spectral shape (Green, 1988). One sound is the standard; the other sound is the signal plus standard. The standard sound often has a flat power spectrum with equal energy per logarithmic frequency unit (equal energy per octave). The signal-plus-standard sound is created by adding energy to some region of the standard spectrum. The sounds are pulsed on for a short duration; 100 to 500 ms is typical. In the earlier experiments, the spectra were collections of sinusoidal components which extended over several octaves in frequency. The power spectrum of the sounds, not the temporal waveforms, is clearly the critical stimulus quantity, because the same thresholds are obtained if the phase for each component is randomly chosen on each presentation (Green and Mason, 1985).

The sounds to be discriminated are presented in random order in two intervals of a forced-choice test. The listener is asked to select the interval containing the signal-plus-standard sound. Each sound is presented at a random level, chosen from a rectangular distribution; the range of the distribution is typically around 20 dB. This variation in overall level discourages the listener from attempting to discriminate between the two sounds on the basis of a *successive* comparison of absolute level in any frequency region. Rather, a *simultaneous* level comparison of different frequency regions within the same spectrum must be made to achieve good discrimination between the two spectra. These spectral discriminations share many essential features with the recognition of a speech utterance, in that the information needed for the discrimination is contained within a single sound presentation, and the phase spectrum is largely irrelevant.

In this paper, we argue that the detection of spectral changes relies on different auditory processes or cues, depending on the bandwidth of the spectrum. That different cue systems must be involved became apparent from the studies of Gilkey (1987) and Kidd (1987). They were the first to present evidence that the sensitivity to changes in spectral shape was not severely impaired if the bandwidth of the sound was greatly reduced. Sensitivity was best for spectra several octaves in width, but it decreased only slightly as the bandwidth of the spectrum was reduced to a few Hertz. The important stimulus quantity is the bandwidth of the spectrum compared to the width of the critical band. A *wide* spectrum extends over more than two critical bands. A *medium* spectrum encompasses one-half to two bands at most. A *narrow* spectrum is less than half the width

of one critical band. Because the size of the critical band changes with center frequency, the number of critical bands encompassed by a fixed-bandwidth sound will change with center frequency. The majority of our research has employed spectra with center frequencies of about 1000 Hz, so most of what we will discuss is limited to that frequency. Recently, we began exploring spectra with very high frequency components, some extending above 8000 Hz. The results obtained with these high-frequency spectra are perplexing, and will be presented after our review of the three different cue systems.

## 2 Wide-Band Spectra

Much of our earlier work on the discrimination of spectral shape involved multitonal complexes that extended over many critical bands. We argued that discrimination of a change in the shape of such spectra involved a simultaneous measurement of the intensity level at different frequency regions of the spectrum and a subsequent comparison of those levels (Green, 1983). Such level comparisons were certainly possible, because the components of the spectrum fell into different critical bands. Each band could then provide an estimate of the level in its frequency region that was largely independent of the estimate provided at some different critical band. The channel model of Durlach, Braida, and Ito (1986) provided a quantitative analysis of such a process, and derived the optimum decision rule for the typical profile experiment. The optimum decision rule has a very simple form when the internal noise (they call it peripheral noise) associated with each channel is much smaller than the random changes in overall level. Because the peripheral noise is about 1-2 dB whereas the change in overall level, an essential control in the profile experiment, is about 20 dB, these conditions are often met. Under these conditions, the optimum decision rule is to compare the level measured in the signal channel with the average level of all the nonsignal channels. Thus, if  $m$  components make up the standard and the signal is an increment in one component, then the optimal decision statistic,  $z$ , is of the following form

$$z = L_s - \frac{1}{(m-1)} \sum_{i \neq s}^m L_i \quad \text{Eq. 1}$$

where  $L_s$  is the level estimated in the signal channel and  $L_i$  is the level estimated in the nonsignal channels. There are  $m-1$  such nonsignal channels. This decision statistic,  $z$ , should be near zero if the standard alone is presented and a large positive value if the signal is added to the standard. Note that such a decision statistic is not influenced by the random changes in overall intensity level, because the value of  $z$  remains the same if the levels of all components are changed by the same amount. Berg and Green (1990) used the COSS (Conditional On a Single Stimulus) procedure to estimate the weights used by listeners in an auditory profile task. COSS, developed by Berg (1989), is a new procedure used to assess the weight that an observer allocates to different components of the stimulus in a discrimination task. For a spectrum extending between 200 and 5000 Hz and containing 5, 7, or 11 components, the estimated weights were very close to the values given by Eq. 1.

While not disputing the mechanism suggested for these wide-band spectra, the research of Gilkey (1987) and Kidd (1987) raised serious questions about the generality of these results. They systematically varied the bandwidth of the spectrum from the more than five-octave range used in the research reported above to a total range of only a few Hertz (much less than  $1/3$  octave). Despite this enormous change in bandwidth, the threshold for detecting spectral change increased only slightly at the very small bandwidths. Clearly, with such narrow-band spectra it is impossible to make independent estimates of the levels of the different components, because they all fall within the same critical band. Other mechanisms must be responsible for the ability to hear spectral changes in these narrower spectra.

### 3 Medium-Width Spectra

Berg, Nguyen, and Green (1992) suggested that for spectra about the width of a critical band the detection of spectral change might be based on a pitch cue. Feth and Stover (1987) were the first to argue that profile analysis might be based on a pitch cue. They applied this idea to spectra extending over several octaves in frequency using the pitch values computed from the envelope-weighted average of the instantaneous frequency (EWAIF). They investigated a variety of conditions, and computed the change in pitch between the standard and signal-plus-standard sounds that were just discriminable. If this idea were successful, then the pitch change should be the same for all conditions. Instead, the changes in pitch value were widely different for the three-component spectra, and only moderately constant for multi-component spectra. The constant value, however, was an order of magnitude larger than what one might expect for a pure-tone stimulus. Richards, Onsan, and Green (1989) showed that a pitch change could not be an important cue for detecting changes in the shape of wide-band spectra because randomizing the rate of the digital-to-analog converter produced little change in threshold.

Berg et al. (1992) used three-tone complexes. The components were all of equal level in the standard sound, and the central component was increased in level to produce the signal-plus-standard sound. The phase of the three components was chosen at random on each presentation, so that the temporal structure of the waveform was unimportant. Because EWAIF calculations are phase dependent, it is better to use the newer procedures to calculate the pitch of a complex sound. These new procedures are called IWAIF (Anantharaman, Krishnamurthy, and Feth, 1993) for intensity or SEWAIF (Dai, 1993) for squared-envelope weighted average of the instantaneous frequency. The chief advantage of these new procedures is that the calculated pitch of a complex narrow-band spectrum is influenced only by the power spectrum of the sound, and is independent of the phase spectrum. Thus, details of the fine structure of the wave are unimportant, and the calculated pitch value is exactly equal to the center of gravity (centroid) of the power spectrum when plotted on a linear frequency scale.

The problem with the theory that pitch mediates the detection of changes in spectral shape is that if the three components are close enough in frequency to fall within the same critical band, then the spectrum is nearly symmetrical in linear frequency. The pitch of such a symmetric spectrum is simply the frequency of the central component of the

complex, and it does not change when the central component is altered in level, because that does not change the symmetry. So how can changing the intensity of the central component induce a pitch change? Berg et al. argued that the listeners based their judgments on the output of a critical band whose center frequency was located at some distance from the central frequency of the complex. Such off-frequency listening (Leshowitz and Wightman, 1971) is known to occur when it will benefit the detection of a signal. In the present case, the advantage is that the spectrum at the output of the filter will be tilted, and thus highly asymmetric, if the critical band is sufficiently remote. For example, if the center frequency of the off-frequency critical-band filter is considerably above the central frequency of the complex, then the lowest frequency tone will be greatly attenuated compared to the central and highest frequency tone. The signal-plus-standard spectrum, by increasing the intensity level of the central component, would lower the center of gravity of the spectrum, and thus decrease the pitch.

Evidence for the existence of such a pitch-detection process was obtained from the COSS weights estimated from the different listeners in this task. For the wider spectra, the weights were nearly -0.5, 1, and -0.5, as would be expected on the basis of Eq. 1. For the narrower spectra, the pattern of COSS weights was quite different. Generally, one edge component had a weight of -1, the weight of the other edge was near zero, and the weight of the central component was about +0.5. The actual pattern of weights depended on the direction of the pitch cue reported by the listener. If the listener said the signal-plus-standard spectrum produced a higher pitch than the standard alone, then the lowest component had a weight near -1. If the listener reported a lower pitch, then the highest component had a weight near -1. Such differences in the direction of the pitch shift can be explained by assuming that the off-frequency critical band was located either above or below the central frequency of the complex. If the off-frequency band were above the center frequency of the complex, the level of the components would systematically increase with frequency. If the off-frequency band were below the frequency of the complex, the level of the components would systematically decrease with frequency. Raising the level of the middle component would in both cases change the pitch, decreasing it in one case and raising it in the other. For the three tones spaced at 920, 1000, and 1080 Hz, two of the four listeners said that the standard was lower in pitch than the signal-plus-standard spectrum, while the other two said the reverse.

While such a mechanism is a plausible account of the detection process for medium bandwidth spectra, it cannot operate for very narrow spectra. For narrow spectra, the components are so close in frequency that off-frequency listening would not produce any noticeable tilt in the spectrum at the output of the critical band. The output spectrum would be nearly symmetric for either an on- or off-frequency critical band. The center of gravity of such a symmetric spectrum, and hence the pitch, will not be appreciably altered by changing the central component of the spectrum. Another mechanism must be proposed to account for the remarkably good detection observed with these narrow-band spectra.

#### 4 Narrow-Band Spectra

Narrow-band spectra have bandwidths less than half the width of a critical band. Investigators have employed spectra that are only a small fraction of the width of a critical band, for example, extending  $\pm 10$  Hz about 1000 Hz. We have argued (Green, Berg, Dai, Eddins, Onsan, and Nguyen 1992) that the detection of changes in the power spectra of such narrow-band sounds is based on changes in the power spectrum of the envelope. Such detection can be accomplished even if the phases of the components that comprise the spectrum are chosen at random on each presentation. The power spectrum of the envelope of a narrow-band random process was first derived by Lawson and Uhlenbeck (1950). For audio power spectra that are rectangular in shape, such as noise or equal-amplitude sinusoids, the power spectrum of the ac part of the envelope is nearly triangular, having a base equal to the bandwidth of the audio power spectrum. Alteration in the flat audio spectrum changes the power spectrum of the envelope in different manners, depending on how the audio spectrum is altered. Figure 1 shows the envelope power spectrum of a standard, equal-amplitude audio power spectrum (solid line) and the envelope power spectrum of three different signal-plus-standard audio spectra (dotted line). In the three panels of Figure 1, a signal is added to the central component of the standard, to one end of the spectrum, or to both ends of the spectrum. In these examples, the signal component(s) is always 5 dB above the level of the other components.

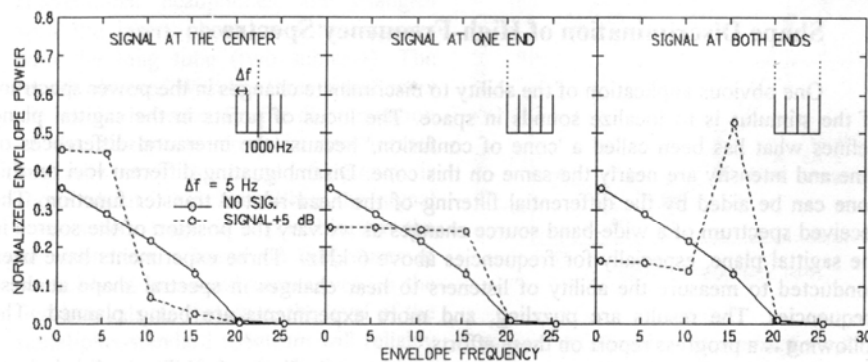


Figure 1. Average envelope power spectra for various audio spectra.

We stress again that these changes in the envelope power spectrum are averaged over different phases for the components of the audio complex, which, of course, produce different envelope waveforms. We argue that envelope power spectra that have a preponderance of low-frequency energy are heard as smooth (see left panel of Fig. 1). If the envelope power spectrum is nearly flat or has a peak at some higher frequency (see middle and right panels of Fig. 1), then the perception is of a rough envelope. In Green et al. (1992), discrimination experiments were conducted to determine the basis for

discriminating between these various narrow-band spectra. The results suggest that the discrimination is based on a single, unidimensional cue system, such as the rough-smooth dimension just discussed.

## 5 Summary

In short, we believe that discriminating changes in spectral shape, profile analysis, depends on three different auditory processes. Each process is most appropriate for a given range of spectral bandwidth. For wide-band spectra extending over several critical bands, the detection mechanism is a simultaneous comparison of the level of different critical bands, such as that suggested in Eq. 1. For medium-band spectra, the detection mechanism is a change in pitch. This pitch value is computed at the output of a critical band located in a frequency region displaced from the central frequency of the sound. The pitch will increase or decrease when the signal is added to the standard spectrum, depending on whether the critical band is above or below the frequency region of the sound's spectrum. For narrow-band spectra, the detection mechanism is a change in the power spectrum of the envelope of the sound. The relative frequency content of the power spectrum of the envelope determines whether the sound is heard as smooth or rough. The sensitivity of all three mechanisms is very similar. Each allows a listener to detect an alteration of about 1-3 dB in the power spectrum of any sound, with changes in a wide-band spectrum being somewhat easier to detect.

## 6 Shape Discrimination of High-Frequency Spectra

One obvious application of the ability to discriminate changes in the power spectrum of the stimulus is to localize sounds in space. The locus of points in the sagittal plane defines what has been called a 'cone of confusion,' because the interaural differences of time and intensity are nearly the same on this cone. Disambiguating different loci on this cone can be aided by the differential filtering of the head-related transfer function. The received spectrum of a wide-band source changes as we vary the position of the source in the sagittal plane, especially for frequencies above 6 kHz. Three experiments have been conducted to measure the ability of listeners to hear changes in spectral shape at these frequencies. The results are puzzling, and more experiments are being planned. The following is a progress report on these efforts.

The goal of the first experiment was to compare the listener's ability to discriminate spectral shape in different frequency regions. Three-tone spectra with medium bandwidths were varied in center frequency  $f_c$  from 250 to 16,000 Hz. For a given center frequency, the lower and higher frequencies were  $f_c/1.38$  and  $1.38 \cdot f_c$ . Thus, all three components presumably fell into different critical bands. For the standard spectrum, all three components were equal in level. The median level of the standard complex was 60 dB SPL per component, and the overall level of the complex was chosen from a rectangular distribution with a 20-dB range. In the signal-plus-standard spectrum, the level of the center component was increased relative to the other two components by adding, in-phase, a component to the central component of the standard. The threshold for this



added signal was determined in a two-alternative forced-choice task using an adaptive two-down one-up procedure whose equilibrium point is 70.7 % correct. The threshold values are reported as the ratio of the added signal to the standard component in decibel,  $20 \log (\Delta p/p)$ .

Because of the problem of standing waves at the higher frequencies, the experiments were conducted using three different sound transducers: (1) a conventional headphone (Sennheiser Headphone, HD-450), (2) an insert earphone (Etymotic ER-2), and (3) a sound delivered through a long (5 meter) lossy plastic tube with a diameter of about 1 cm. The subjects listened binaurally with the phones in phase for the conventional headphone. For the other conditions, the listening was monaural. The long lossy tube provides a nearly perfect impedance match at the entrance of the ear canal, and, by decreasing reflections at this point minimizes changes in pressure level as a function of frequency at the eardrum (Zhou and Green, 1994). For some of the higher frequency conditions, some nonlinear distortion products were audible, thus for  $f_c > 5$  kHz, a lowpass noise band was used extending below 4 kHz, and having a spectrum level of 24 dB.

Figure 2 shows the mean results obtained with three normal-hearing subjects. The signal-to-standard ratio at threshold is plotted as a function of the center frequency  $f_c$ . The circles represent the thresholds obtained with the conventional headphones, the triangles with the insert phone, and the squares with the long tube (two subjects). The results obtained with the three transducers are nearly the same. Profile discrimination of the three-tone complex deteriorates markedly once the center frequency is above about 5000 Hz. When the signal level is very large, it is possible that the discrimination is based on differences in overall level measured successively in the two intervals. In that case, the level of the signal-plus-standard spectrum will reliably exceed the level of the standard spectrum, despite the 20-dB randomization of presentation level. The dotted line shows the threshold value achievable for an ideal detector listening to only changes in overall level during the two intervals of the forced-choice task. Many of the measured thresholds exceed this level. We interpret this to mean that the listeners continued to listen for the simultaneous changes in spectral shape, even when the successive differences between overall level provided a potentially stronger cue. No attempt was made to instruct the observers to listen to the successive changes in overall level.

Two more experiments were conducted in the free field, which provides an

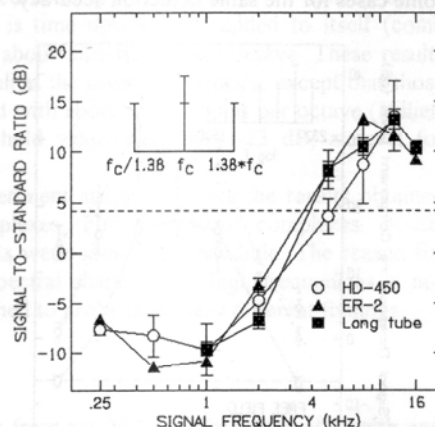


Figure 2. Profile discrimination for a 3-tone complex.



experimental setting very similar to that used to study sound localization. In the first free-field experiment, we used a stimulus similar to that used in the headphone experiment, except that the number of components was increased from three to seven. The seven components were spaced in frequency at equal intervals on a logarithmic scale extending from 2327 to 16,000 Hz. Each of the seven components was set at a constant sensation level based on absolute threshold measurements for each individual listener. The median sensation level of the entire complex was about 60 dB, and the overall level was randomly varied  $\pm 5$  dB about that level.

Figure 3 presents data for this experiment. Each symbol represents data for a single listener. The dotted line again represents the performance possible for an ideal observer listening only to successive changes in absolute level. The observed thresholds nearly always exceed these levels. These data are similar to those obtained in the earlier headphone experiments. The general result is that listeners are not able to hear changes in the spectral shape of a multiple component complex at the higher frequencies. In the speech range, listeners can hear 1-dB changes in the intensity of a single component of a seven-component complex. At the higher frequencies, a change of over 20 dB is needed in some cases for the same detection accuracy.

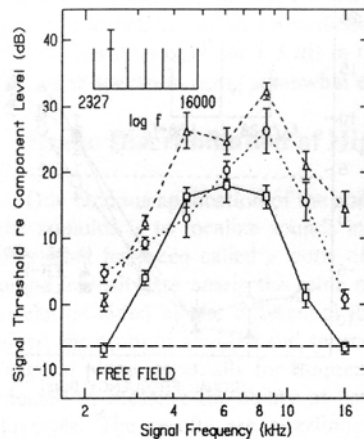


Figure 3. Profile discrimination at different frequencies of a 7-tone complex.

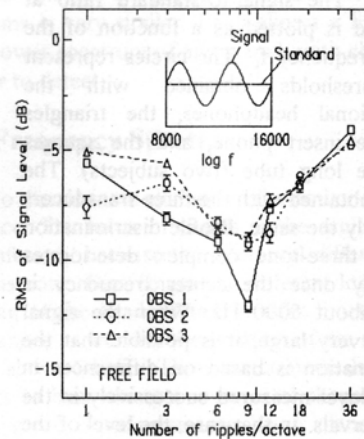


Figure 4. Profile discrimination with rippled noise at different ripple frequencies.

In the second free-field experiment, a very different stimulus was used to measure the ability to hear changes in spectral shape--what we call a rippled-noise stimulus. A nearly continuous spectrum was employed that sounded noise-like rather than tonal. The standard spectrum contained many equal-amplitude components spaced in frequency 6.25 Hz apart (the reciprocal of the duration). The phase of each component was randomized for each presentation. The components extended in frequency over the range from 8 to

16 kHz. The change in spectral shape was created by imposing a sinusoidal ripple on that flat spectrum. The ripple was sinusoidal on a logarithmic frequency scale. The phase of that ripple was randomized on each presentation to discourage detection based on the absolute level of the spectrum in any frequency region. The size of the ripple was adjusted adaptively in a three-down one-up forced-choice procedure, which produces an equilibrium point of 79.4% correct. We can think of the signal-plus-standard spectrum as the addition of a set of 'signal' components to the standard spectrum. Each 'signal' component is either in- or out-of-phase with the corresponding component in the standard spectrum, thus producing an increase or decrease in the level at that frequency. The threshold can be expressed as the rms of this 'signal' compared to the rms level of the standard components. The loudspeaker was digitally equalized to produce a standard having a flat amplitude spectrum, as measured with a sound level meter (Bruel and Kjaer Model 1613) located at the position of the listener in the sound-attenuated chamber.

Figure 4 shows discriminability data for this rippled-noise stimulus. The ordinate is the threshold rms signal-to-standard ratio in decibels. The abscissa is the number of ripples per octave. All three listeners were able to discriminate changes in the spectral shape of this noise-like stimulus. The listeners reported hearing a difference in the quality of the noise similar to that heard with noise that is time delayed and added to itself (comb filtering). The best detection occurred with about nine ripples per octave. These results are similar to those obtained for similar stimuli at the lower frequencies, except that those studies found that the best detection occurred with about three ripples per octave (Hillier, 1991; Green, 1993). Also the lowest threshold value was about -23 dB, at least for wideband spectra, 100-6400 Hz.

These results of the rippled-noise experiment are quite unlike the results obtained with the three- or seven-component complexes. For multi-tonal complexes, 30-dB changes in the level of individual components were sometimes inaudible. The reason for these differences in sensitivity changes in spectral shape at the high frequencies is not understood, and further experiments are planned to probe these very different findings.

## 7 Acknowledgments

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### Comment by P.W.J. van Hengel

As you showed in figure 4 of your manuscript, your subjects have a threshold minimum for detection of rippled noise at a ripple frequency of about 9 ripples per octave. If this ripple frequency is "translated" to distance along the cochlea using an exponential place-frequency map it corresponds to a periodic spatial distribution with a periodic of approximately 0.4 mm. As I have shown in my presentation (van Hengel, this volume) this distance follows from our cochlea model as an ideal separation of SOAE's that is also found in measurements.

Do you think there might be a relationship between the phenomena?

**Reply**

This is an interesting speculation. Unfortunately the common finding at *lower* frequencies ( $f < 8000$ ) is that the minimum occurs at 3 ripples per octave rather than 9 ripples per octave, for example Hillier (1991) and Green (1993).

In short, for there to be a real relationship, the periodic spatial distribution should be *wider* at lower frequency regions of the cochlea - which is inconsistent with SOAE data.