Knowles Hearing Center
Northwestern University

Contemporary Hearing Science Inspired by David M. Green

SPEAKER ABSTRACTS

Transition Bandwidths and the Cadence Model

Bruce G. Berg, Ph.D.

Associate Professor, Cognitive Sciences
School of Social Sciences
University of California, Irvine

Transition bandwidth refers to a robust phenomenon that provides a unique view of the dynamics of automatic auditory attention. Transition bandwidths are observed in band-widening experiments in which listeners detect an increase in the amplitude of the central tone in a background of equal-intensity tones. Spectral cues are degraded with roving level and roving-frequency procedures. As stimulus bandwidth is increased by adding tones with a constant frequency separation, Δf, thresholds increase up to a certain bandwidth and then decrease sharply. The peak of the threshold function presumably marks a transition from a process of temporal envelope discrimination to a process akin to profile analysis. The transition occurs automatically, presumably when the stimulus becomes wide enough to support across channel comparisons. This suggests that the ten-fold range in estimated transition bandwidths might arise from differences in listeners' abilities to parse a spectrally dense stimulus into different comparison channels. A challenging finding is that the general shape of the threshold function is largely independent of Δf for values ranging from 10 Hz to 400 Hz. A cadence model, which uses simulated period histograms as input, provides a quantitative description of thresholds and decision weights when the stimulus bandwidth is less than the transition bandwidth.

Going Green? Overview of the linear-systems extension of auditory profile analysis using maximum-density carriers.

David A. Eddins, Ph.D., CCC-A

Departments of Communication Sciences & Disorders, Chemical & Biomedical Engineering,
Director, Auditory & Speech Sciences Laboratory
University of South Florida

Research on spectral shape perception using the auditory profile analysis approach was spear-headed by David M. Green and his colleagues during the 1980s to mid-1990s. This work included relatively simple stimuli including standard stimuli consisting of a multi-tone complex with components spaced on a logarithmic scale and signal stimuli produced by incrementing the level of one of those components. Several investigations used multi-component signals, including increments and decrements in multiple components resulting in sinusoidal spectral shapes (e.g., Bernstein and Green; JASA 82, 1587-1592, 1987). Towards the end of this period, several investigators extrapolated this line of research to a broader class of stimuli in a paradigm commonly referred to as spectral modulation detection or spectral ripple detection (the latter term should not be confused with ripple in the passband of a filter or iterated rippled noise). Like the early

profile analysis research, investigations of spectral modulation detection have used a variety of carrier densities from sparse (~ 5 components/octave) to maximum density (limited by sample frequency and duration or number of samples). This review of spectral modulation detection will begin with the influence of basic stimulus parameters, such as modulation frequency, modulation waveform shape, carrier bandwidth, carrier frequency region, carrier density, overall level, and overall duration. More advanced topics will include indices of tuning to spectral modulation and the potential impacts of hearing loss and advancing age on spectral modulation detection. Finally, several potential clinical applications of such a measure will be discussed.

Signal Detection Theory and Psychophysics in the Real World

Lawrence L. Feth, Ph.D.

Professor, Department of Speech and Hearing Science
Ohio State University

The publication of "Signal Detection Theory and Psychophysics" (Green and Swets, 1964) heralded the wide-spread adoption of n-interval forced-choice methods in psychoacoustics. Levitt's (1971) "Transformed Up-Down Methods in Psychoacoustics" shortened data collection time substantially and made "2IFC, 2-Up, 1-Down adaptive-tracking" the first choice for many studies conducted over the past five decades. Thus, when my lab group was asked to help with studies of the audibility of various aircraft in a variety of real soundscapes "2IFC, 2-Up, 1-Down" was the "obvious" paradigm to use. The initial study tested the detectability of two different helicopters in nine different sound environments. Listeners were asked to report which of two intervals contained a helicopter sound, and then to identify which of the two helicopters they heard. Analysis of the detection data was difficult; the identification data proved to be impossible to interpret. This presentation reports two studies that benefitted from the problems encountered in the helicopter study. First, we tested the predictions of a computation model based on the results of the helicopter study. Recordings of three different fixed-wing aircraft were used as targets to be detected in the same nine soundscapes used previously. The computational model and our human listeners were required to "process" the same recorded sound files. To better approximate real listening conditions, the 2IFC adaptive procedure was replaced by the SIAM (Single Interval Adjustment Matrix, Kaernbach, 1990) procedure. Careful calibration of the experiment to ensure that both human listeners and the computational model processed the same input was essential to providing a fair test of the model. Results showed surprisingly good agreement between model predictions and human performance. The second study investigated the use of feedback in detection studies. The 2IFC paradigm and trial-by-trial feedback are used to minimize response bias and deliver a criterion-free index of sensitivity. However, trial-by-trial feedback is unrealistic in real-world listening situations. In realistic listening situations, feedback may be possible for some signal-plus-noise conditions, but not for others. This study analyzed the effects of incomplete feedback on listener bias in a simple tone-in-noise detection task. Bayesian Cognitive Modelling was used to analyze the data for conditions ranging from no feedback, to feedback for Signal presentation trials, Yes response trials, Correct Response trials, to full feedback. We maintain that both careful calibration of recorded signals and more realistic experimental designs are necessary to apply TSD to real world listening tasks, but useful results are possible.

Adaptive Plasticity of Loudness: Evidence and Clinical Relevance

Craig Formby, Ph.D.

Distinguished Graduate Research Professor
Department of Communicative Disorders
University of Alabama

Over a half-century period, from the early 1940's to the early 1990's, clinicians focused on therapeutic strategies using moderate- and high-level sound to attack two of the most intractable audiological conditions, namely, limited sound tolerance and tinnitus. In 1991, Hazell & Sheldrake (Proceedings of the Fourth International Tinnitus Seminar, pp. 245-248, 1992) described a novel sound-therapy application of low-level broadband noise to treat tinnitus patients, who also suffered clinically significant hyperacusis. Their successful application of low-level sound therapy to treat hyperacusis, although little noted at that time, was historic, providing a unique insight into the adaptive plasticity of the auditory system, specifically, the adaptive plasticity of loudness. In this presentation, I will review both basic and clinical evidence of the adaptive plasticity of loudness, including our controlled studies in which we manipulated the plasticity of loudness in applications of low-level sound therapy to expand the auditory dynamic ranges of typical hearing-impaired persons. This promising treatment approach offers audiologists a new clinical tool to treat sensorinueral hearing loss, making possible unique opportunities to enhance hearing-aid benefit with less reliance on wide dynamic range compression. [Research supported by NIDCD]

Study of auditory attention in a signal detection task through selection of stimulus cues that best ameliorate the effects of signal uncertainty

Ervin R. Hafter, Ph.D.

Professor
Department of Psychology
University of California, Berkeley

Selective attention refers to enhanced processing of an attended source in the presence of others. While the choice may be between domains such as auditory and visual, or auditory and thinking about lunch, it can also affect performance within a single domain such as auditory frequency when the signal is a tone in the presence of a wide-band noise. In studies of sensory processing, selective attention has been described in terms of internal filters that separate stimuli along the dimension of interest with judgments based on only the band that contains the signal. For this to be optimal, the observer must know the right band(s) to monitor, an argument often studied in experiments that examine the effects of "signal uncertainty." Seminal work on the problem of frequency uncertainty (Green, 1961) examined psychometric functions to determine the relation between stimulus level and frequency uncertainty. Following Dave's lead, our lab used a variety of stimulus cues to alleviate uncertainty in hope of describing filters for the internal dimension on which psychoacoustic judgments were made. In a case of tonal signals in noise, direct measures of the internal bandwidths of the filters showed they were not fixed, as for auditory filters in cochlea, but rather grew with uncertainty. In another case, where subjects detected 3-tone complexes whose components were either unrelated or harmonically related; uncertainty was alleviated by cues that pointed either to a frequency-based dimension or one based on the derived dimension, musical pitch.

Binaural and Monaural Edge Pitch

William M. Hartmann, Ph.D.

Professor
Department of Physics and Astronomy
Michigan State University

In 1966 David Green published a JASA article entitled "Signal detection analysis of equalization and cancellation model" ({40}, 833-838) which differed in several respects from the well-known model by Nat Durlach. Specifically Dave envisioned a process whereby the binaural system would subtract OR ADD the left and right channels. Years later Mark Klein and I (1980) discovered the binaural edge pitch which had an intrinsic ambiguity that could only be understood by assuming that the binaural system can both subtract and add the two channels. The case for both subtraction and addition operations was made stronger by work with Colleen McMillon (2001). This talk will review the data and modeling of these binaural pitch effects and make the connection to recent work on the monaural analogs.

A dual-channel, spectrotemporal model of pure-tone frequency discrimination

Haunping Dai, Ph.D.

Associate Professor
Department of Speech, Language and Hearing Sciences
The University of Arizona

The purpose of this presentation is to further develop a dual-channel model for pure-tone frequency discrimination [Dai, H. Nguyen, Q., and Green, D.M. (1995). Hear. Res., 85, 109-114.] In this model, a change in the frequency is detected based on a relative change in the output levels of two auditory filters that sandwich the tone. The filters are centered far enough from the tone that, on the filter skirts where the tone sits, the slopes are relatively steep. This detection scheme improves the predicted frequency JND by cancelling noise terms (either external or internal) that are common to, or correlated between, the pair of filters or channels. To be explored here is an idea that the internal noise associated with the two channels has a common term due to the temporal synchrony in the neural responses between the two channels. As such, the model predicts that the JND worsens (grows larger) towards higher frequencies due to the increasingly diminishing synchrony in the neural responses. The model predictions and implications will be discussed.

Contributions of Specific Frequency Bands to the Loudness of Broadband Sounds

Walt Jesteadt, Ph.D.

Director, Psychoacoustics Laboratory
Center for Hearing Research
Boystown National Research Hospital

I became interested in loudness while working with Dave Green as a postdoc at Harvard, where loudness research was part of the local culture. For the past 5 years, I have been using procedures developed by other presenters at this meeting to obtain perceptual weights showing the contributions of specific frequency bands to the loudness of broadband sounds. Data obtained with tonal and noise-band stimuli show greater weight assigned to the low- and especially high-frequency edges of broadband stimuli than current loudness models would predict. Data obtained with a loudness matching procedure show the high-frequency edge effect, but not the low-frequency edge effect. The pattern of weights in listeners with hearing loss is altered by amplification, but does not change over the first three months of hearing aid use. Current loudness models provide a good account of peripheral processes contributing to loudness. The edge effects and the increased loudness of tonal stimuli compared to noise stimuli may both be due to central processes.

From profile analysis to the cocktail party problem: methods and insights inspired by signal detection theory

Gerald Kidd, Jr., Ph.D.

Professor, Director, Psychoacoustics Laboratory
Department of Speech, Language and Hearing Sciences and Hearing Research Center
Boston University

Current work in our laboratory aims to better understand performance in solving "the cocktail party problem" in which the listener must understand the spoken message of one talker in the midst of competing talkers and other unwanted sources of sound. The contemporary approaches we use in studying this complex problem trace back to the author's experience working with Dave Green on the series of studies termed "auditory profile analysis" and even further back to Green's early work on signal detection theory and uncertainty. This talk will trace the history of the ideas and supporting evidence that led from profile analysis to current work on the cocktail party problem. Of particular interest is the importance of listener expectation established through a priori knowledge and experience. The influence of expectation is pervasive and is found in such diverse examples as talker identification and message recognition at low signal-to-noise ratios and algorithms that rely on exact knowledge of the signal in performing sound source segregation.

Projects with David

Dennis McFadden, Ph.D.

Ashbel Smith Professor Emeritus
Center for Perceptual Systems
University of Texas

Over the years, I served with David Green on two major projects – the reenactment of the JFK assassination in Dealey Plaza, and a committee under the Ocean Studies Board charged with evaluating the effects of intense sounds on marine mammals. Recollections will be shared, some accurate.

Signal Detection Theory in Perception and Physiology: Good for What Ails You

John C. Middlebrooks, Ph.D.

Professor
Department of Otolaryngology
University of California, Irvine - School of Medicine

Work in our laboratory addresses the general hypothesis that perception is in some way a product of brain activity. Signal detection theory provides us with a common language for expressing perceptual and neural performance in humans and other animals. With varying degrees of success, we attempt to relate perceptual algorithms, the software, to neural mechanisms, the wetware, with performance expressed by the sensitivity index, d'. I will describe some recent and ongoing research: (1) Enhanced restoration of hearing with a penetrating auditory nerve electrode; (2) Spatial stream segregation in human and cat psychophysics and cortical physiology; and (3) Non-invasive physiological measures of frequency and temporal acuity.

Sensitivity to envelope coherence, revisited

Brian C. J. Moore, FMedSci, FRS

Department of Experimental Psychology,
University of Cambridge, England

The ability to discriminate amplitude modulation (AM) from frequency modulation (FM) imposed on a sinusoidal carrier, when the AM and FM are equally detectable, may depend on two types of cues: (1) The FM is associated with fluctuations in instantaneous frequency (IF) while the AM is not (a cue based on temporal fine structure, TFS); (2) fluctuations in excitation level on the two sides of the excitation pattern are in phase for AM and out of phase for FM (referred to as the AM-phase cue). It has been shown that AM-FM discrimination, with equally detectable AM and FM, is better for a 2-Hz than for a 10-Hz modulation rate. This might reflect greater sensitivity to TFS for low than for high rates. Alternatively, it might reflect a worsening of AM-phase discrimination with increasing rate. Green, Richards and Onsan (JASA, 1990) assessed discrimination of the phase of AM applied to two sinusoidal carriers, but they did not use rates below 4 Hz. Here, AM phase discrimination was assessed using rates from 2 to 20 Hz. A band of noise centered between the two carriers was used to prevent use of changes in IF at the outputs of auditory filters centered between the two carriers. Young and older subjects with normal hearing were tested. Performance was almost constant for AM rates from 2 to 10 Hz, but worsened at 20 Hz. Performance was near chance for AM depths near the detection threshold. The results suggest that the superior AM-FM discrimination at 2 Hz cannot be explained in terms of comparison of the phase of fluctuations on the two sides of the excitation pattern, but rather depends on the use of TFS cues.

Authors:
Brian C. J. Moore
Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, England
Aleksander P. Sęk
Institute of Acoustics, Faculty of Physics, Adam Mickiewicz University, Poznań, Poland

Sound Source Localization Under Real-World Conditions: Multiple and Moving Sound Sources and Moving Listeners

William A. Yost, Ph.D.

Research Professor
Department of Speech & Hearing Sciences
College of Health Solutions
Arizona State University

Dave Green published many papers dealing with sound source localization (including a chapter Dave and I co-authored, Binaural Analysis, Handbook of Sensory Psychology, Springer, 1975). The review of sound source localization co-authored by Dave and John Middlebrooks (1991, Annual Reviews of Psychology, 42) is still one of the most cited reviews of sound source localization. This talk will briefly describe some of the results we have obtained in the Spatial Hearing Lab at ASU regarding three related funded projects dealing with sound source localization in conditions that occur in the real world (e.g., in a sound field): 1) Auditory Processing of Multiple Sound Sources Producing Nearly Simultaneous Sound: The Size of the Auditory Scene: Our research to date suggests that the auditory scene (the Cocktail Party) is small, i.e., fewer than 4-5 sound sources. And, spatial separation among sound sources makes, at best, a small difference in the size of the auditory scene. Results from two of the paradigms we have used will be briefly described: numerosity judgments and spatial release from masking (SRM). 2) Sound Source Localization by Rotating Normal Hearing Listeners and/or Sound Sources: Results from a thorough evaluation of the Wallach Azimuth Illusion (WAI, based on the work of Hans Wallach, 1940, Journal of Experimental Psychology, 27) will be briefly described. The WAI is a robust illusion providing a powerful way to evaluate how auditory-spatial and head-position cues are integrated to allow for world-centric sound source localization. The WAI occurs for a particular scenario of sound source and listener rotation in the azimuth plane. 3) Sound Source Localization by CI Patients (in collaboration with Michael Dorman's CI Laboratory): CI patients with microphones not in their concha or ear canal have a large number of front-back reversals (FBRs, i.e., azimuthal front-back errors) in localizing high-frequency sounds, when normal hearing listeners do not. However, these FBRs are substantially reduced when the CI patients rotate their heads much in the same way normal hearing listeners can disambiguate FBRs for low-frequency sounds when they rotate their heads. [research funded by NIH (R01DC015214–WAY and F32DC017676 –WAY and MTP), Facebook Reality Labs (WAY and MTP), Advanced Bionics Corporation (MD), and MED EL Corporation (MD)]

Authors:
William A. Yost and M. Torben Pastore, Spatial Hearing Laboratory, College of Health Solutions, Arizona State University

The Wald sequential test as a statistical criterion for the signal level change in adaptive staircase up-down procedures

Jan Zera

Faculty of Electronics and Information Technology
Warsaw University of Technology, Poland

Staircase up-down adaptive procedures used for the measurement of sensory thresholds have several advantages, such as a simple rule for signal level setting and relative robustness to fluctuations of the subject's attention during the measurement. A restriction of those procedures is that they allow to estimate the threshold level only for a small number of points on the psychometric function and the estimation of each point requires a different decision rule for signal level changing. To overcome this limitation a decision rule based on the Wald sequential statistical test, similar to that applied in the PEST adaptive method [Taylor and Creelman, JASA 41, pp. 782-787 (1967)], is proposed is this study. The Wald test is simple to implement and allows to target any point on the psychometric function by entering adequate input parameters. Numerical simulations have shown that the adaptive up-down staircase procedure combined with the Wald test for signal level setting well reproduces the tracks of the usual version of this procedure and may estimate the sensory threshold with similar accuracy, and similarly low bias and random error levels. Numerical simulations were supported by an exemplary experiment in which masked thresholds were measured for human subjects. The results indicated that the modified procedure, with a decision rule based on the Wald test, produced methodologically proper results when it was used to target both the points estimated with the usual adaptive up-down staircase procedure and also other points on the psychometric function.