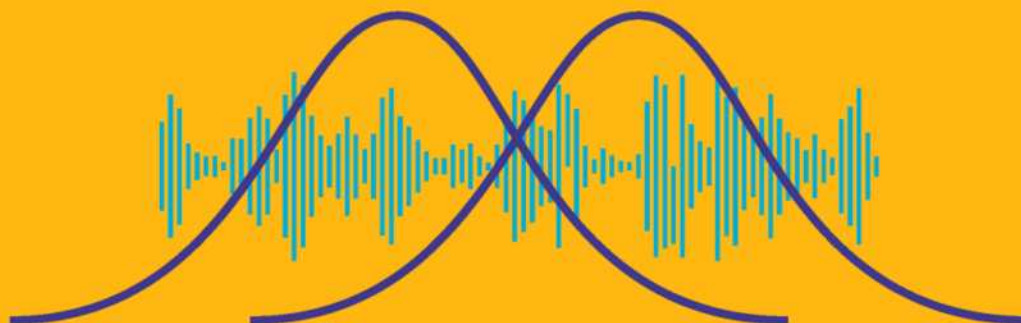


Knowles Hearing Center at Northwestern University



Contemporary Hearing Science inspired by David M. Green

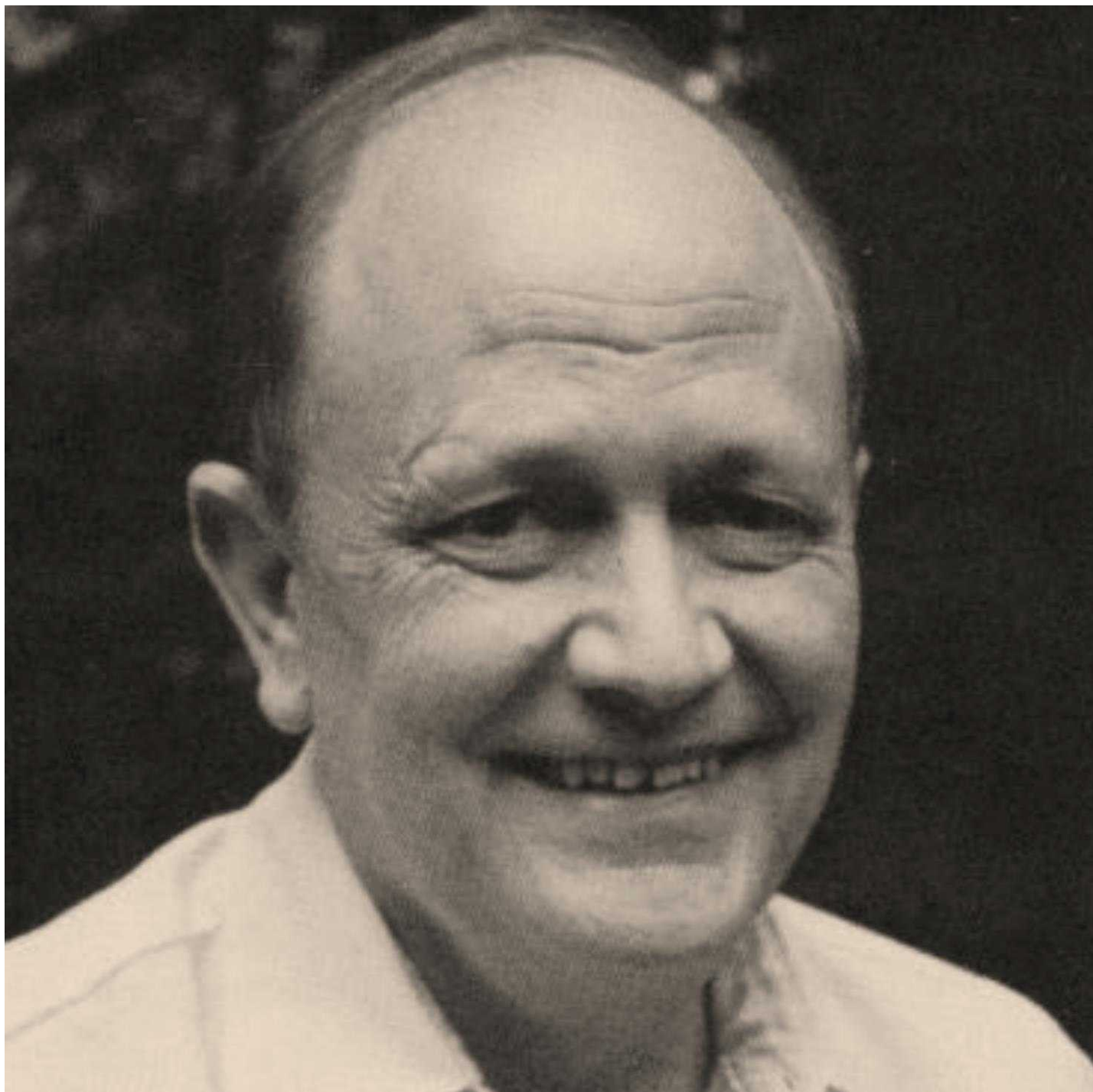
July 25 and 26, 2019

Special guest: David M. Green
Northwestern University, Evanston, Illinois

Details and registration to be posted soon at knowlesmeeting.northwestern.edu

Speakers include

Bruce Berg <i>Univ. of California, Irvine</i>	Craig Formby <i>University of Alabama</i>	Jungmee Lee <i>University of South Florida</i>	Bob Shannon <i>Univ. of Southern California</i>
Les Bernstein <i>University of Connecticut</i>	Erv Hafter <i>Univ. of Calif., Berkeley</i>	Bob Lutfi <i>University of South Florida</i>	Beth Strickland <i>Purdue University</i>
Huanping Dai <i>University of Arizona</i>	Bill Hartmann <i>Michigan State University</i>	Dennis McFadden <i>Univ. of Texas, Austin</i>	Bev Wright <i>Northwestern University</i>
Dave Eddins <i>University of South Florida</i>	Walt Jesteadt <i>Boys Town Nat. Res. Hosp.</i>	John Middlebrooks <i>Univ. of California, Irvine</i>	Bill Yost <i>Arizona State University</i>
Larry Feth <i>Ohio State University</i>	Gerald Kidd <i>Boston University</i>	Brian Moore <i>Univ. of Cambridge</i>	Jan Zera <i>Warsaw Univ. of Tech.</i>



David M. Green

CITATION FOR DAVID M. GREEN

...for contributions to knowledge, theory, and methodology in audition.

CAMBRIDGE, MASSACHUSETTS • 8 JUNE 1994

DAVID MARVIN GREEN was born in Jackson, Michigan on 7 June 1932. He received B. A. degrees from The University of Chicago in 1952 and The University of Michigan in 1954. In an introductory psychology course at Michigan he was assigned to a section taught by Wilson P. Tanner, Jr., from the vantage point of the theory of signal detectability. Dave was recruited to serve as a subject in John Swets's thesis experiment on human signal detection and then in his first year of graduate school in psychology he joined Tanner, Swets, and T. G. Birdsall in auditory detection research at the Electronic Defense Group, a laboratory of the electrical engineering department. He spent his third postgraduate year with J. C. R. Licklider at the Massachusetts Institute of Technology (MIT), returned to Michigan for a year to receive his Ph.D. in 1958, and then returned to MIT as assistant professor of psychology. He began his consulting affiliation with Bolt Beranek and Newman Inc. (BBN) in 1958, which continues. He went on to associate and full professorships in four prominent psychology departments: University of Pennsylvania, 1963–66; University of California at San Diego, 1966–73; Harvard University as professor of psychophysics and 3 years as department chair, 1973–85; and since then, the University of Florida as graduate research professor.

While an undergraduate in Ann Arbor, Dave married Clara Lofstrom. Their four children are Allan, Phillip, Katherine, and George, now grown and active in various fields. Clara died in 1978 of cancer. Dave and Marian Heinzmann were married in 1980. They now enjoy two grandchildren, summers in a cabin in Maine, and weekends on a boat near their home in Florida.

Dave's scientific contributions span the basic topics in the psychology of hearing. His M. A. thesis explored the relationship of signal intensity and duration. His doctoral work was on multiple-component and noise signals. In between, he co-authored technical reports applying signal detection theory to audition and speech recognition. He wrote an invited tutorial article on detection theory for JASA two years later. Fundamental contributions continued on the topics of uncertain signal frequency, sequential observations and speed-accuracy tradeoffs, critical bands, internal noise, masking with tones and noise, sound localization, temporal acuity, and frequency discrimination. With R. D. Luce he published a series of important papers on signals presented at random intervals, response times, magnitude estimation, and absolute identification. Apart from work on such substantive problems, he made many lasting contributions to theory and method in psychophysics and related fields. He co-authored *Signal Detection Theory and Psychophysics* in 1966 and wrote *An Introduction to Hearing* in 1976.

In the past 10 years, with many colleagues in his laboratory, in some 35 articles and a 1988 book, Dave has made a major contribution in an area he termed "profile analysis." This work has shown that people listen to patterns across the frequency spectrum even when set to detect changes at a single frequency and, moreover, that changes in a tone are better detected when it is presented along with others. This concept countered the ideas of selective attention and auditory filters that concentration on a single frequency channel is most effective and is facilitated by presenting a tone in isolation.

Dave's applied research also makes important contributions. He has published about 15 articles on several subjects with several colleagues at BBN, notably with Sanford Fidell on environmental noise. He mixes basic with applied research at BBN; preparation of the book *Signal Detection Theory and Psychophysics* was supported under a (NASA) contract there. One

might imagine that his more-than-80 articles in JASA constitute nearly his full set of publications, but he has published a like number of articles and chapters elsewhere, primarily in the psychology literature. His productivity seems to be increasing (from about 3 publications a year earlier in his career to about twice that many recently) and so we can look forward to the many good things that lie in store for us.

Dave is an uncommonly good citizen of the scientific community. He has been associate editor of JASA and consulting editor for other journals, chairman of the society's committee on psychological and physiological acoustics, president of the society, member of the National Institutes of Health (NIH) study sections, member of several committees of the National Research Council (NRC), and three-time chairman for a total of 5 years of the NRC's Committee on Hearing, Bioacoustics, and Biomechanics (CHABA).

The scientific community has expressed its appreciation of David Green, both as scientist and colleague, in many ways. He received the Biennial Award (now the Lindsay Award) and the Silver Medal from this society. He was elected a fellow of the Society of Experimental Psychologists and received its Howard Crosby Warren Medal. He is a fellow of the American Psychological Association and recipient of its Distinguished Scientific Contributions Award. He is a fellow of the American Academy of Arts and Sciences and is a William James Fellow of the American Psychological Society. He was a fellow of St. Johns College, Cambridge, England and of All Souls College, Oxford, England. He is a member of the National Academy of Sciences. To go with his two B. A. degrees, the University of Cambridge, England, and Harvard University granted him M. A. degrees.

Dave joined the ASA during his graduate years, after Licklider dropped in on Tanner, Swets, and Green at Michigan to see what was going on in signal detection theory there and introduced them to the society. Dave was immediately exposed to what must have been the world's all-time greatest group of informal mentors. People like Licklider, Egan, Jeffress, Neff, Kryter, Hirsh, and Pollack greeted such newcomers, commented helpfully on their presentations, and invited them to meetings of the committee on psychological and physiological acoustics, at which formal business vied with intense debates about burning technical issues through half the night. As he matured in the society, Dave most visibly took on this unusually generous interest in the beginners' growth and recognition. He regards them all as having their stories to tell—and after a few years of his tutelage, they really do. As might be expected, a steady stream of young people joined his laboratory for graduate study or post-doctoral years, to receive the full Green treatment. They watched an open and relentlessly inquiring mind sort through issues with astonishing clarity. They left ingrained with the ideas that overcoming the obstacles in research was the most fun and that they could just go ahead and do things they might not think they could. His former students form a solid core of their field. Although his own research sets him apart, it is widely appreciated that his willingness and ability to give the best of mentorship in his laboratory and in the society is one of Dave's outstanding professional traits and one of the main reasons for thinking of him as so highly deserving of the society's Gold Medal.

JOHN A. SWETS

Reimaged from Dave's ASA Gold Medal Citation JASA 1994

Brief Homily on Signal Detection Theory

David M. Green

I would like to give a brief homily on signal detection theory (SDT). As is illustrated in the poster for this event, the iconic image for SDT is two Gaussian density functions separated by about one standard deviation. I will argue that, while this icon is commonly used, it is inappropriate for illustrating the essential contribution of the theory. Rather, we should be focusing on more general measurements of a signal's detectability and how those different measurements are related. (I must disclose that I had nothing to do with the advertisements that promoted this event nor, for that matter, the choice of the honoree.)

In determining how human observers detect weak signals, historically, we often simply asked them whether they heard, saw, or sensed a given signal. Such responses are private evaluations and there is no means of counting them as anything more than individual opinions. The responses do not indicate whether or not the signal was actually detected. The sensations produced by the stimuli are subjective; they are private or covert. The only objective fact is simply the observer's response on that particular trial. Signal detection theory (SDT) suggests that many trials should be presented with the signal present on some and with it absent on others. Estimating the proportion of times the observer says he detects a signal when none is present can provide a more objective measure of the signal's detectability than simply estimating the times he says yes to signals that are always presented. The proportion of affirmative responses when the signal was not presented (false alarms) as well as the proportion of affirmative responses when the signal was presented (signal detections) provides a more useful data set. If the observer is persuaded to be stricter or more liberal in how much information is necessary for the subject to commit to a 'yes' response, then an entire collection of different points can be obtained. A single point is obtained when one plots the hit proportion on the ordinate and the false-alarm proportion on the abscissa. The collection of such points is called a receiver-operating characteristic curve (ROC curve). The different points represent different criteria for those subjective events that the observer is willing to call a 'signal.' If the observer could not detect any of the signals, the

point for that block of trials would fall on a straight line where the hit-rate and false-alarm rate are equal to one another (the chance line). If the observer was able to detect the signal, then the hit rate should be larger than the false alarm rate. The ROC curve represents how detectable the signal is (how far the curve exceeds the chance line). Except for highly unusual cases, the ROC curves begin at the lower left corner of the graph when the false-alarm rate is quite low, rise monotonically over the middle range of values of false-alarm rate, and then asymptote into the upper right corner of the graph when the false-alarm rate approaches 1.0.

SDT assumes that the different stimulus alternatives (signal or non-signal) produce different sensory events. The collection of different sensory events is characterized by different density functions. One such density function is labeled the signal density $f_s(x)$ and the other density is labeled nonsignal density $f_n(x)$. We assume that a larger the value of x has a greater likelihood of representing a signal event rather than a nonsignal event. The observer establishes a criterion C , and if the sensory events exceed C then the observer says yes. Thus, if the signal is present, the probability that the sample exceeds C yields the hit probability $P(Y|s)$. But, if the nonsignal event is presented and the sample exceeds C , we have the false-alarm probability $P(Y|n)$. These two probabilities are the two coordinates for each point on the ROC curve,

$$P(Y|s) = \int_C^{\infty} f_s(x) dx \quad (1)$$

$$P(Y|n) = \int_C^{\infty} f_n(x) dx \quad (2)$$

Another detection procedure, advocated by many engineers, is called the A/B test. In this test, a pair of stimulus alternatives is presented. The target (A) and nontarget (B) alternatives are presented in temporal succession or in different spatial locations, and the observer is forced to choose the correct order. In addition, this procedure is often called a two-alternative forced-choice (2AFC) task, because the two intervals are presented successively in time. The observer is asked to indicate which of the two intervals contain the signal.

Signal detection theory can also describe the A/B test. Let us call the A alternative a signal event and the B alternative a non-signal event. In the A/B task, what produces a correct response? The observer gets two samples, one from the signal alternative and one from the nonsignal alternative. A correct response will occur if the sample, x_s is

greater than the nonsignal sample x_n . That will occur if a sample from f_s is greater than the sample from f_n , that is, the sample presented in the nonsignal interval. Or to express it in integral terms,

$$P(C) = P(x_s > x_n) = \int_{-\infty}^{\infty} f_s(x) \int_{-\infty}^{x_s} f_n(y) dy dx \quad (3)$$

The probability of being correct is simply that integral when summed for all possible values of x_s .

{I should note here for extra credit, that if, in Eq. 3, you raise the second integral to the M-1 power you have the probability of being correct in an M interval forced-choice task. The nonsignal samples are all independent, and hence all M-1 samples must be less than x_s and that joint probability is simply that probability P (raised to M-1 power). I will comment more about this matter when we discuss the relation to the ROC curve.}

Next, we relate these terms to the probability in a yes-no task $P(Y|s)$ and $P(Y|n)$. The inner term of Eq. 3 is simply the complement of the false-alarm probability at the given value of x_s , that is, $1 - P(Y|n)$ at x_s ,

$$1 - P(Y|n)|_{x_s} = \int_{-\infty}^{x_s} f_n(y) dy \quad (4)$$

This is true because the entire integral of $f_n(x)dy$ summed from minus infinity to plus infinity is equal to one.

Now consider the fundamental theorem of calculus. Suppose we have a function of y given x , $y = f(x)$. The theorem states that the increment in area dA under the curve y is simply equal to the value of y multiplied by the increment in x ,

$$dA = y dx \quad (5)$$

That increment in area dA is simply a rectangle with y as the height and dx as the base.

To relate this fact to the area under the ROC curve, think of interchanging the abscissa and ordinate of the ROC curve. Let the hit rate run along the abscissa and the false-alarm rate run along the ordinate. As in the calculus example, the change in the area dA under the ROC curve is simply the complement of the false-alarm rate ($1 - P(Y|n)$) times the increment of the probability of a hit $P(Y|s)$ at that point. The small rectangle representing the change in area dA has height $1 - P(Y|n)$ and a base $P(Y|s) + dx$. So the total area under the ROC curve is equal to the probability of being

correct in a two-alternative forced-choice task. This assertion certainly makes sense in the extreme. The chance line of the ROC curve $P(Y|s) = P(Y|n)$ yields 50% correct in a two-alternative forced-choice task, whereas if the ROC curve covers the entire square the percent correct is 100%. (For those seeking added credit, prove that the chance line of an M-alternative ROC curve yields $1/M$ as the percent correct in an M-alternative forced choice task.)

{Again for extra credit, note that if you raise the complement of the false-alarm rate $1-P(Y|n)$ to the M-1 power before measuring the area, you have the relation between the new area and the percent correct in an M-interval forced-choice task.}

The crucial point is that the relationship between the area under the ROC curve and the percent of correct decisions in an A/B test does not depend on any assumption about the form of the probability densities f_s or f_n . They need not be Gaussian; they could be any probability functions whatsoever. We do assume that as x increases in value, there is an increasing likelihood that x is a signal event rather than a nonsignal event. This assumption makes the observer a likelihood-ratio observer, one whose performance is known to be optimum in communication science. Note also that where the density functions can take on any form, optimal performance can be evaluated in terms of Kublick-Leibler divergence (D_{KL}), also known as information divergence or discriminable information¹,

$$D_{KL}(f_s||f_n) = E_{f_s} \ln \left(\frac{f_s}{f_n} \right) \quad (6)$$

D_{KL} is the expected value of the log-likelihood ratio and reduces to the square of d' where the density functions are equal-variance, normal.

The moral of this homily is, that while individual detection responses are covert subjective quantities, there are procedures that can convert these individual responses to completely objective data. These data are as objective as any of the quantities used in the so-called hard sciences. The contribution of signal detection theory is that the analysis provides a means of understanding the structure of these different detection tasks and predictions about how the quantities measured in each task should be related. This analysis assumes the existence of probability density functions. But these densities need not be Gaussian; that is only on one of many possible assumptions.

We should comment, for those who work in the field, that an ROC curve with a $d'=1$ (roughly the case illustrated in the icon promoting this event) corresponds to 76%, a proportion nearly half way between chance and perfect performance in a two-alternative forced-choice task. But, please note that the area under the ROC curve being equal to the percent correct in the 2AFC does not depend on the value of d' or any other parameter of the two hypothesized densities.

So, when we are planning the 2029 version of this conference, I would suggest using the two Gaussian distributions plus an ROC curve with the area under the curve darkened and printing in large white letters on that area “76%.”

This manuscript was enormously improved by the editorial comments of Prof. D. McFadden and Prof. R. Lutfi and Marian H. Green.

¹ Kullback, S., and Leibler, R. A. (1951). “On information and sufficiency,” *Ann. Math. Stat.* 22, 79–86.

Knowles Hearing Center
Northwestern University

Contemporary Hearing Science Inspired by David M. Green

Thursday July 25

- 8:45 am *Introduction*
Beverly Wright; Northwestern University
- 9:00 am *Signal Detection Theory and Psychophysics in the Real World*
Lawrence L. Feth; Ohio State University
- 9:30 am *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; University of California, Berkeley
- 10:00 am *Going Green? Overview of the linear-systems extension of auditory profile analysis using maximum-density carriers.*
David A. Eddins; University of South Florida
- 10:30 am *Signal Detection Theory and the Inverse Problem in Audition*
Robert A. Lutfi; University of South Florida
- 11:00 am *Adaptive Plasticity of Loudness: Evidence and Clinical Relevance*
Craig Formby; University of Alabama
- 11:30 pm Message from Dave Green
David M. Green
- 12:00 pm **BREAK FOR LUNCH**
- 1:30-3:00 **POSTER PRESENTATIONS**
- 3:00 pm *Binaural and Monaural Edge Pitch*
William M. Hartmann; Michigan State University
- 3:30 pm *From the Ear to the Brain – From Dave Green to Cochlear Implants*
Robert V Shannon; University of Southern California
- 4:00 pm *Changing the Channel*
Beth Strickland; Purdue University
- 4:30 pm *Auditory Perceptual Learning*
Bev Wright; Northwestern University
- 5:00 pm *A dual-channel, spectrotemporal model of pure-tone frequency discrimination*
Huanping Dai; University of Arizona

Friday July 26

- 9:00 am *Contributions of Specific Frequency Bands to the Loudness of Broadband Sounds*
Walt Jesteadt; Boystown National Research Hospital
- 9:30 am *Evidence of possible contribution of cochlear mechanics to auditory perception from studies of otoacoustic emissions*
Jungmee Lee; University of South Florida
- 10:00 am *From profile analysis to the cocktail party problem: methods and insights inspired by signal detection theory*
Gerald Kidd, Jr.; Boston University
- 10:30 am *Projects with David*
Dennis McFadden; University of Texas-Austin
- 11:00 am *Signal Detection Theory in Perception and Physiology: Good for What Ails You*
John C. Middlebrooks; University of California, Irvine - School of Medicine
- 11:30 am *Sound Localization of Rifle Shots*
Dennis McFadden on behalf of Dave Green
- 12:00 pm **BREAK FOR LUNCH**
- 1:30-3:00 pm **POSTER PRESENTATIONS**
- 3:00 am *Combining cross-correlation and signal-detection theory approaches to account for the binaural abilities of normal-hearing listeners and listeners with "slight" hearing loss*
Les Bernstein; University of Connecticut
- 3:30 pm *Sensitivity to envelope coherence, revisited*
Brian C. J. Moore, University of Cambridge, England
- 4:00 pm *Sound Source Localization Under Real-World Conditions: Multiple and Moving Sound Sources and Moving Listeners*
William A. Yost; Arizona State University
- 4:30 pm *The Wald sequential test as a statistical criterion for the signal level change in adaptive staircase up-down procedures*
Jan Zera; Warsaw University of Technology, Poland
- 5:00 pm *Transition Bandwidths and the Cadence Model*
Bruce G. Berg; University of California, Irvine

POSTERS

- 1 *Use of Signal Detection Theory to Quantify the Effects of Adaptive Nonlinear Frequency Compression on S-SH Confusions*
Joshua M. Alexander; Purdue University
- 2 *In Search for the Measures of Temporal Fine Structure Sensitivity*
Agudemu Borjigin & Hari Bharadwaj; Purdue University
- 3 *Profile Analysis without Excitation Patterns: Neural Fluctuation Profiles*
Laurel H. Carney; University of Rochester
- 4 *Estimating the Duration of a Refractory Period in Perceptual Learning*
Alex Clain & Beverly A. Wright; Northwestern University
- 5 *Masking and negative masking of low-level signals by low-level maskers under fixed and roved conditions*
Christopher Conroy & Gerald Kidd, Jr.; Boston University
- 6 *The Precedence Effect: Is it the same for interaural time and interaural level differences (ITDs and ILDs)?*
Raymond Dye; Loyola University of Chicago
- 7 *Spectral weighting for sound localization in the free field.*
Monica L. Folkerts; Vanderbilt University Medical Center
G. Christopher Stecker; Boys Town National Research Hospital
- 8 *Age-related differences in ITD-related binaural interference and interference release*
Yuan He & Jennifer J. Lentz; Indiana University
- 9 *Analysis of different methods and parameters for generating spectro-temporal modulation: a comparison of behavioral thresholds and model predictions*
Sittiprapa Isarangura¹, Ann C. Eddins¹, Frederick J. Gallun² and David A. Eddins¹: ¹University of South Florida; ²VA Portland Health Care System
- 10 *Development of duration discrimination during adolescence*
Gay JD, Rosen MJ and Huyck JJ; Kent State University
- 11 *Learning Interference from Sequential but Not Interleaved Training on Two Perceptual Tasks: Evidence That Anterograde Learning Interference Arises from an Interaction of Tasks in Different Learning Stages*
Ruijing Ning; Northwestern University
- 12 *Synergy of Spectral and Spatial Segregation Cues in Simulated Cocktail Party Listening*
Brianna Rodriguez, Robert Lutfi and Jungmee Lee; University of South Florida
- 13 *Further exploration for cochlear contributions to individual differences in cocktail-party listening*
John Sheets¹, Jungmee Lee¹, Joshua Hajicek², Glenis Long³ & Robert Lutfi¹; ¹University of South Florida; ²University of Michigan; ³CUNY
- 14 *Typical speech exposure during fetal auditory neurodevelopment*
Hannah M. Smith & Brian B. Monson; University of Illinois at Urbana-Champaign
- 15 *Access to phonetic information at extended high frequencies improves speech-in-speech performance*
Allison Trine & Brian B. Monson; University of Illinois at Urbana-Champaign
- 16 *Singing Proficiency in children with bimodal hearing and bilateral cochlear implants*
Jing Yang, Rosalie Uchanski, Emily Hahn, Xianhui Wang, Lisa Davidson, & Li Xu; University of Wisconsin- Milwaukee, Washington University, Ohio University

Knowles Hearing Center

Northwestern University

Contemporary Hearing Science Inspired by David M. Green

SPEAKER ABSTRACTS

Thursday July 25

8:45 am

Introduction

Beverly Wright

Professor

Department of Communication Sciences and Disorders, Northwestern University

9:00 am

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.

9:30 am

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.

10:00 am

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

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

Professor

Departments of Communication Sciences & Disorders, Chemical & Biomedical Engineering, 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.

10:30 am

Signal Detection Theory and the Inverse Problem in Audition

Robert A. Lutfi, Ph.D.

Professor

Department of Communication Sciences and Disorders, University of South Florida

Few single works have had as great an impact on contemporary hearing science as Dave Green's book *Signal Detection Theory and Psychophysics* co-authored with John Swets. In that book Dr. Green explains how information in signals defined statistically can be used to identify a class of ideal detectors that broadly serve as a standard for evaluating limits in human sensitivity across different stimulus conditions and psychophysical tasks. The present talk focusses on the application of this approach to understanding how we identify rudimentary properties of objects and events from the sounds they make – the so-called inverse problem. Studies are reviewed wherein the human listener performing in the task is modelled as an ideal detector limited by internal noise and an inability to use information in higher moments of the probability densities of acoustic features distinguishing sound sources. The conclusion of these studies is that basic limits in auditory sensitivity exert a far greater influence on listener judgments than does knowledge of the intrinsic acoustic relations that would serve to disambiguate sound sources. [Research supported by NIDCD R01-DC001262]

11:00 am

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 sensorineural hearing loss, making possible unique opportunities to enhance hearing-aid benefit with less reliance on wide dynamic range compression. [Research supported by NIDCD]

11:30 am

Message from Dave Green

David M. Green Ph.D.

12:00 noon BREAK FOR LUNCH

1:30-3:00 POSTER PRESENTATIONS

3:00 pm

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" (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.

3:30 pm

From the Ear to the Brain – From Dave Green to Cochlear Implants

Robert V Shannon

Research Professor

Caruso Department of Otolaryngology-HNS, University of Southern California

What would it be like to have electroreception like a shark? Or echolocation like a porpoise? Can a Martian see purple? Dave green asked such questions on his exams to challenge the students to apply what they know about perception to a new type of sensory system that they could not experience. Such training proved to be useful for working with cochlear implants (CIs). Electrical stimulation of the auditory nerve produces sound sensations, but the pattern of nerve activity is totally unnatural. In the beginning many people were misled by thinking of implant hearing as being similar to acoustic hearing, which led them down many false paths. Instead it is useful to think of implant hearing not as acoustic hearing, but as if the recipients were sharks or Martians. Psychophysics can tell us the basic capabilities at detecting and discriminating but doesn't tell us how that information is integrated into complex patterns. In implants we have a brain that was trained by a lifetime of auditory experience, now confronted by a pattern of neural information that is quite unnatural. What will it sound like? What patterns of neural information are most important for understanding speech? Music? Sounds in nature? Surprisingly, most psychophysical characteristics look similar between electric and acoustic stimulation, once we correct for the loudness. Speech can be understood with only broad patterns of activation modulated in time. This kind of information is provided by cochlear implants and allows patients to converse on the telephone. Fine timing and spectral information, however, are not provided by a CI and we can see that they are critical for harmonic pitch, music and emotion in speech.

Cochlear implants are allowing us to see the synergy between the basic aspects of sound and their importance in understanding complex patterns of sound. Now if we ever encounter a Martian we may have a model for understanding their sensory capabilities, thanks to cochlear implants.

4:00 pm

Changing the Channel

Beth Strickland, Ph.D.

Professor

Department of Speech, Language and Hearing Sciences, Purdue University

A great deal of early psychoacoustic research focused on the importance of a critical band, which might reflect a single channel in the auditory system. A surprising finding by Green and colleagues in the 1980s was that performance on intensity discrimination tasks improved if components were added which should be well outside the critical band. This type of task was called “profile analysis”, and showed that listeners seemed to be comparing information across channels to make decisions. In one of the early papers on profile analysis [Green, Mason, and Kidd; JASA 75, 1163-1167, 1984] Green and colleagues found that thresholds improved with duration of the profile. One hypothesis that they explored was the idea that the system had some type of “automatic gain control” that took about 50 ms to activate. Although this hypothesis did not seem to fit the data pattern in that paper, the idea of gain adjustment within a channel in response to sound has since received considerable attention. This talk will review evidence for within-channel gain adjustment.

4:30 pm

Auditory Perceptual Learning

Beverly Wright, Ph.D.

Professor

Department of Communication Sciences and Disorders, Northwestern University

Performance on many perceptual tasks improves with practice, indicating that our sensory systems are not rigid but rather can be changed through experience. My coworkers and I have been investigating the factors that induce and those that prevent perceptual learning on auditory skills, including how those factors change with age and are affected by sensory and cognitive disorders. Conclusions drawn from learning on fine-grained auditory discrimination tasks have held for visual and speech learning, suggesting that common principles are at play across multiple domains. Knowledge of these issues will lead to more effective perceptual training strategies to aid rehabilitation and promote skill enhancement.

5:00 pm

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

Huanping 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.

Friday July 26

9:00 am

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.

9:30 am

Evidence of possible contribution of cochlear mechanics to auditory perception from studies of otoacoustic emissions

Jungmee Lee, Ph.D.

Research Associate Professor
Department of Communication Sciences and Disorders, University of South Florida – Tampa

Measures of otoacoustics emissions (OAEs) provide increasing evidence of possible contributions of cochlear mechanics to human auditory behavioral tasks. Detection thresholds are better at the frequencies of spontaneous OAEs (e.g., Long & Tubis, 1988; Lee & Long, 2012); Amplitude modulation perception is correlated with amplitude modulation of $2f_1$ - f_2 , distortion product OAEs recorded with amplitude-modulated f_1 and steady-state f_2 (Lee & Dhar, 2013); Decision weights in two-tone level discrimination are correlated with levels of stimulus frequency OAEs (SFOAEs) of the tones (Lee et al., 2015). Recent pilot data from our lab suggest that internal noise efficiency calculated using COSS analysis (Berg, 1990) in a cocktail-party listening task is related to signal-to-noise ratio of SFOAEs.

10:00 am

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

Gerald Kidd, Jr., Ph.D.

Professor

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.

10:30 am

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.

11:00 am

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

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.

11:30 am

Sound Localization of Rifle Shots

A reading from Dennis McFadden on behalf of Dave Green

12:30 pm BREAK FOR LUNCH

1:30-3:00 pm POSTER PRESENTATIONS

3:00 pm

Combining cross-correlation and signal-detection theory approaches to account for the binaural abilities of normal-hearing listeners and listeners with "slight" hearing loss

Les Bernstein

Professor

Departments of Neuroscience and Surgery (Otolaryngology), University of Connecticut

Among the numerous contributions David M. Green made to psychophysics in general and psychoacoustics in particular was his explication and application of Signal Detection Theory (SDT). In this presentation, several experimental contexts will be discussed in which a SDT approach to modeling has yielded successful and intuitively appealing accounts of measures of binaural auditory processing. The primary focus will be on recently published empirical data and quantitative modeling from our laboratory. Those reports demonstrate that data obtained in binaural detection experiments conducted across the last five decades can be accounted for by combining a signal-detection-based decision variable with a cross-correlation-based model of binaural processing that incorporates stages of peripheral and central auditory processing. Notably, some of the data obtained in those experiments had remained either theoretically unaccounted for or, at best, had been only accounted for via ad hoc approaches. Key to the development of the unified account of those experimental results was 1) the inclusion of "internal noise" within the stages of the model and 2) the calculation and inclusion of the variability of the interaural correlations of the outputs of the model for both masker-alone and signalplus-masker conditions. Empirical data and quantitative modeling will also be presented that demonstrate how the SDTinspired approach has also proven useful in determining and explaining why some listeners with slight, but clinically negligible, elevations in audiometric thresholds exhibit reliable and meaningful deficits in both binaural detection and binaural discrimination tasks. [Supported by Office of Naval Research (N00014-15-1-2140; N00014-18-1-2473)]

3:30 pm

Sensitivity to envelope coherence, revisited

Brian C. J. Moore, FMedSci, FRS

Department of Experimental Psychology, University of Cambridge, England

Aleksander P. Sęk

Institute of Acoustics, Faculty of Physics, Adam Mickiewicz University, Poznań, Poland

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.

4:00 pm

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, Arizona State University

M. Torben Pastore

Faculty Research Associate

Department of Speech & Hearing Sciences, 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)]

4:30 pm

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 in 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.

5:00 pm

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.

POSTER ABSTRACTS

1

Use of Signal Detection Theory to Quantify the Effects of Adaptive Nonlinear Frequency Compression on S-SH Confusions

Joshua M. Alexander

Department of Speech, Language and Hearing Sciences, Purdue University

Frequency lowering in hearing aids can cause listeners to perceive [s] as [ʃ]. The S-SH Confusion Test (Alexander 2019, JSLHR) was designed to help clinicians and researchers document these negative side effects. Listeners' responses on this test can be readily analyzed using concepts from signal detection theory. A systematic investigation of the perceptual effects of the newest method of frequency lowering, adaptive nonlinear frequency compression (ANFC), was conducted on 61 listeners [36 normal-hearing (NH), 25 hearing-impaired (HI)]. Listeners were divided into 3 groups whereby speech was processed with 8-9 different FL settings appropriate for mild-to-moderate, moderately-severe, or severe-to-profound hearing loss. d-prime for the original method (NFC) was similar to Alexander (2019) and was surprisingly better than ANFC. Results suggest that performance with ANFC becomes poorer with an increasing mixture of lowered and un-lowered audio. Whereas, NFC was primarily associated with false alarms ([ʃ] for [s]), ANFC was associated with lower hits ([ʃ] for [s]) and even higher false alarms, thereby shifting the bias towards [s] relative to Alexander (2019). The patterns of d-prime across conditions were similar for the HI and NH listeners, however bias for HI listeners was consistently more negative (biased for [ʃ]) across all conditions.

2

In Search for the Measures of Temporal Fine Structure Sensitivity

Agudemu Borjigin, and Hari Bharadwaj

Purdue University

While the temporal fine-structure (TFS) in low-frequency sounds can convey information about pitch and spatial location, a nuanced debate exists about the role of these TFS cues in masking release. The long-term goal of the present study is to leverage individual differences to understand the role of TFS in everyday hearing. As a first step, we sought to measure TFS sensitivity from a large pool of normal hearing listeners through both behavioral and electrophysiological approaches.

Behaviorally, we observed large individual differences in monaural frequency modulation (FM) and binaural interaural time difference (ITD) detection tasks. Moreover, individual differences in these two measures correlate with each other, suggesting that monaural TFS coding can be a primary bottle neck determining binaural sensitivity. The correlation, however, drops significantly after attention score from each measure being factored out. It suggests that the correlation partially reflects non-sensory factors such as attention and motivation.

Electrophysiologically, we passively measured TFS-ITD sensitivity and monaural frequency following response (FFR). A couple of EEG features such as evoked response latency correlate well with behavioral ITD thresholds, while there is no correlation with FM thresholds. It indicates that monaural FM detection is a poor assay of TFS sensitivity, while these two EEG measures and ITD thresholds could be better indicator of TFS coding.

Profile Analysis without Excitation Patterns: Neural Fluctuation Profiles

Laurel H. Carney

Departments of Biomedical Engineering and Neuroscience, University of Rochester

Green's Profile Analysis (1988) provides a solution to several challenges for the power spectrum model; this approach takes advantage of within-stimulus differences in responses across frequency channels, instead of relying on across-stimulus differences in the responses of single channels. Profile analysis extends naturally from tasks such as masked detection of tones to representations of complex sounds. However, profile analysis is generally assumed to depend on energy-based representations across frequency channels, and thus suffers from the limited ability of auditory-nerve (AN) fiber average rates to encode stimulus spectra. Excitation patterns based on average discharge rates of AN fibers deteriorate as sound level increases, especially when based on high-spontaneous-rate (HSR) AN fibers or on the responses of cell types in the cochlear nucleus that provide the major inputs to higher auditory centers. These cells, like HSR AN fibers, have low thresholds and limited dynamic ranges. One solution to the limitations of rate-based excitation patterns is the profile of low-frequency fluctuations in responses of AN fibers. Neural fluctuations excite or suppress the auditory CNS, such as midbrain neurons that are tuned to low-frequency fluctuations. Neural fluctuation profiles are shaped by inner-hair-cell transduction saturation, and convey a normalized, inverse representation of spectral amplitude information.

Estimating the Duration of a Refractory Period in Perceptual Learning

Alex Clain and Beverly A. Wright

Department of Communication Sciences and Disorders, Northwestern University

Auditory skills improve with practice. Producing across-day learning requires a sufficient number of trials per session; additional trials in the same session produce no additional improvement. Thus, trials appear to integrate within a session to reach a learning threshold, after which there is a refractory period during which trials do not contribute to learning. The refractory period duration must be <24 hours because training on consecutive days yields learning each day, but its exact duration is unknown. To narrow the range of possible durations, we trained young-adult, normal-hearing listeners on a basic auditory task (interaural-level-difference discrimination) either for two sessions of sufficient training separated by 30 minutes (n=10) or 10 hours (n=10), or for just one session (n=15), and assessed their learning on the next day. The 30-minute group learned no more than the single-session group, indicating the second session did not aid learning, whereas the 10-hour group learned more than the other two groups, indicating the second session provided additional benefit. These results suggest that the refractory period lasts >30 minutes, but <10 hours, and does not require sleep to reset. Knowledge of the refractory period duration will improve the efficiency of perceptual training for typical and clinical populations.

Masking and negative masking of low-level signals by low-level maskers under fixed and roved conditions

Christopher Conroy & Gerald Kidd, Jr.

Department of Speech, Language & Hearing Sciences and Hearing Research Center, Boston University

A peculiar finding in near-“threshold” psychophysics is that, under certain conditions, the detectability of a low-level signal can improve with the presentation of a low-level masker, a finding termed “negative masking” [NM; Raab, Osman, & Rich, JASA, **35**, 1053 (1963)]. One explanation of NM, known as the channel uncertainty hypothesis (CUH), is based on a theoretical interpretation of signal detectability originally proposed by Green. The CUH posits that NM reflects a reduction—via cueing by the masker—of the “large initial uncertainty” [Green, JASA, **33**, 897 (1961) p. 903] that limits the detectability

of low-level signals in quiet. This study examined certain predictions of the CUH. In two experiments, the detectability of a low-level, 1-kHz sinusoidal signal was measured in the presence of a low-level, 1-kHz sinusoidal masker under both fixed and roved conditions. The rove was intended to introduce uncertainty relative to fixed conditions and was hypothesized to mitigate NM. In one experiment, the signal and masker were both 100-ms, were gated simultaneously, and the masker was roved in level. In a second, the signal was 40-ms, the masker was 500-ms, and the signal was roved in time. Uncertainty had an effect, but its magnitude depended on the rove dimension. Results are discussed in terms of the CUH and with respect to Green's continued influence on this area of research.

6

The Precedence Effect: Is it the same for interaural time and interaural level differences (ITDs and ILDs)?

Raymond H. Dye, Jr. and Sarah E. Darnell
Loyola University of Chicago

Until recently, it was believed that the precedence effect operated for both ITDs and ILDs. Stecker and Brown (2010) demonstrated the absence of precedence effects for ILDs under conditions producing strong effects for ITDs using a paradigm in which 16 Gaussian pulses either increased (OR) or decreased (RO) in magnitude of the binaural cue. At short interclick intervals (ICI=2 ms), threshold ITDs were higher for OR than for RO with the difference decreasing as the ICI was lengthened. For ILDs, little difference was found between RO and OR thresholds at any of the ICIs tested (2 – 10 ms), suggesting an absence of onset dominance for ILDs. In order to provide a quantitative measure of the precedence effect, echo weights were measured using a correlational analysis (Richards and Zhu, 1994; Lutfi, 1995) as a function of echo delay for both ITDs and ILDs in 14 individuals. Stimuli were pairs of 3000-Hz 4-ms Gaussian pulses presented with echo delays between 4 ms and 96 ms. Echo weights at short echo delays (4 ms) were significantly higher for ILDs ($M=0.445$) than ITDs ($M=0.196$). We speculate that the mechanism by which ILDs are extracted (integration of sound pressure over time) makes later non-zero ILDs informative in that they signal new sound sources entering the acoustic environment.

7

Spectral weighting for sound localization in the free field.

Monica L. Folkerts¹ and G. Christopher Stecker²

¹Department of Hearing and Speech Sciences, Vanderbilt University Medical Center

²Boys Town National Research Hospital

The ability to localize sound in azimuth is due to the timing and level differences between both ears, interaural-time-differences (ITDs) and interaural-level-differences (ILDs). The duplex theory of sound proposes that low frequencies are localized based on ITDs and high frequencies are localized based on ILDs [Strutt 1907, Philos Mag 13:214- 32; Macpherson & Middlebrooks 2002, J Acoust Soc Am 111:2219-36]. Low frequency ITDs have been indicated as the cue predominately used during sound localization [Wightman & Kistler 1992, J Acoust Soc Am 91:1648-61]. As for the specific frequencies utilized in sound localization, supporting evidence has suggested a frequency "dominance region" around 750-Hz [Bilsen & Raatgever 1973, Acustica 28:131-132]. In order to directly quantify the frequencies utilized during sound localization we adapted techniques to measure temporal-weighting-functions [Stecker and Hafter, 2002, JASA 112:1046-57] and measured spectral-weighting-functions (SWFs) which quantify the relative weights across frequency. Listeners were asked to localize 100-ms complex tones containing seven components or noise bursts containing seven sub-bands. Multiple linear regression of the azimuthal response estimated the SWF weights. Presenting all components/sub-bands at equal sound levels resulted in increased weights for frequencies within the "dominance region". Presenting components/sub-bands at different sound levels resulted in altered weights. [Supported by NIH R01-DC016643]

Yuan He and Jennifer J. Lentz

Department of Speech and Hearing Sciences, Indiana University

Interaural time differences (ITDs) are critical to sound source localization and facilitate the understanding of speech in noise. Unfortunately, the ability to use these cues to lateralize sounds declines with age, even in the absence of hearing loss. The present study further probes the effects of age and central auditory processing on the perception of ITDs in young, middle-aged, and older listeners. We measured ITD-discrimination thresholds for a target narrow-band signal centered at 2.1 kHz in the presence and absence of a concurrent narrow-band interferer at 250 Hz. The interferer was presented either diotically or dichotically with a random ITD chosen over a range of ± 600 microseconds. We also measured a release from interference by cueing the ITD of the interferer using precursors identical to the interferer. Preliminary results suggest that middle-aged and older listeners require larger ITDs than younger listeners to lateralize the target in the presence of the interferer. Diotic interferers had little effect on the ITD thresholds of young listeners but increased thresholds more for middle-aged and older listeners. All age groups benefitted from pre-cueing the interferer. Notably, deficits in the middle-age group were similar to those in the older group.

Sittiprapa Isarangura¹, Ann C. Eddins¹, Frederick J. Gallun² and David A. Eddins¹¹Department of Communication Sciences and Disorders, University of South Florida-Tampa²VA RR&D National Center for Rehabilitative Auditory Research, VA Portland Health Care System

One approach to better understanding temporal and spectral envelope perception is to measure sinusoidal modulation detection thresholds. To date, investigations of temporal modulation (TM), spectral modulation (SM), and spectro-temporal modulation (STM) have used different signal generation methods that may have influenced measurement outcomes. Specifically, the choice of stimulus parameters such as carrier type (e.g., tone complex or noise) and modulation waveform shape (i.e., sinusoidal on linear-log or log-log scales) are potentially important considerations. The purpose of this study is: 1) to compare behavioral TM, SM, and STM detection thresholds across different signal generation methods and different stimulus parameters; and 2) to use a model of auditory peripheral and cortical processing to predict the behavioral detection thresholds. In Experiment I, detection thresholds were measured for eight stimulus conditions (TM at 4 and 32 Hz, SM at 0.5 and 2 cycles/octave, STM at a combination of those four modulation frequencies) and compared across five different stimulus generation methods. In Experiment II, the NSL Auditory model (nsltools) developed by the Neural Systems Laboratory was used to predict the detection thresholds for the stimulus conditions of Experiment I. The results showed that: 1) the different signal generation methods have a modest effect of detection thresholds; and 2) the model as implemented here can successfully predict detection thresholds in some but not all conditions

Gay JD, Rosen MJ and Huyck JJ

Kent State University

Temporal processing, which is important for comprehending speech, matures over an extended developmental period. Here we investigated duration discrimination during adolescence. Listeners aged 8-19 years (four age groups) heard three broadband noises on each trial, and indicated which, if any, of the noises was different in length (longer or shorter). Noise durations were set at 15, 30, 50, 100, or 200 ms. During each of two sessions on consecutive days, each combination of stimulus durations (e.g., 15 vs 100 ms) were presented a total of twenty times in pseudo-randomized order, including identical comparisons (e.g., 100 vs 100 ms) to enable calculation of false alarm rates. All age groups showed a higher (better) sensitivity (d') for comparisons whose durations were more different from one another (e.g., 15 vs. 200 ms). Nevertheless, duration discrimination abilities continued to develop up to age ~14 years: 8- to 10-year-olds did not differ from 11- to 13-year-olds, but 11- to 13-year-olds had lower sensitivity than 14- to 17-year-olds and young adults, who themselves did not differ from one another. Thus, even simple duration discrimination abilities may continue to develop into adolescence.

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Perceptual skills improve with practice, providing a means to treat perceptual disorders and to enhance normal perceptual abilities. However, learning on one perceptual task can be disrupted by training on another. A greater understanding of this interference could lead to more optimal training strategies. We recently observed that training on interaural-time-difference (ITD) discrimination disrupted improvement on interaural-level-difference (ILD) discrimination in the anterograde [ITD-ILD] but not retrograde [ILD-ITD] direction. The specificity of this interference to the anterograde case suggests that the interference occurs during the earliest stage of learning on ILD, the encoding stage. Here, we tested the hypothesis that such anterograde interference occurs because an influence of training on the first task lingers in the encoding stage, altering the encoding of the second task. If anterograde interference occurs simply because of a non-beneficial mixture of the two tasks in the encoding stage, then alternating training between the two tasks, while both are still clearly being encoded, should also disrupt the learning. To test this prediction, we examined learning on ILD (two-interval-forced choice; standard: 4 kHz, 0 dB ILD) between a training session and a testing session the next day. Four groups of young, normal-hearing adults were trained with different regimens: 1) 300 trials of ILD, 2) 300 trials of ILD preceded by 300 trials of an interaural-time-difference discrimination task (ITD) (standard: 0.5 kHz, 0 microsecond ITD), and 3) 300 trials of ILD and 300 trials of ITD alternating every 60 trials. Practicing only ILD generated learning on ILD. This learning was disrupted when practice on ITD was completed before practicing ILD (anterograde interference), but not when practice alternated between ILD and ITD every 60 trials. These results suggest that the anterograde interference of training on ITD on learning of ILD does not occur simply because the influence of the ITD training lingers in the encoding stage. Rather, it appears that the training on ITD must enter a different learning stage, namely consolidation, by sufficient training, before it becomes disruptive to the encoding of ILD. The implication is that the initially encoded form of ITD learning is modified during consolidation into another form and it is this consolidated form that is disruptive to learning on ILD.

Synergy of Spectral and Spatial Segregation Cues in Simulated Cocktail Party Listening

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An approach is borrowed from Measurement Theory [Krantz et al. (1971). Foundations of Measurement, Vol.1] to evaluate the interaction of spectral and spatial cues in the segregation of talkers in simulated cocktail-party listening. The goal is to determine whether mathematical transformations exist whereby the combined effect of cues can be additively related to their individual effects. On each trial, the listener judged whether an interleaved sequence of 4 vowel triplets (heard over headphones) were spoken by the same BBB_BBB... or different ABA_ABA... talkers. The talkers had nominally different fundamental frequencies and spoke from nominally different locations (simulated using Kemar HRTFs). Natural variation in these cues was simulated by adding a small, random perturbation to the nominal values independently for each vowel on each trial. Psychometric functions (PFs) relating d' performance to the difference in nominal values were obtained for the cues presented individually and in combination. The results revealed a synergistic interaction of cues wherein the PFs for cues presented in combination exceeded the simple vector sum of the PFs for the cues presented individually. The results are discussed in terms of their implications for possible emergent properties of cues affecting performance in simulated cocktail-party listening. [Supported by NIDCD R01-DC001262].

Further exploration of contributions to individual differences in cocktail-party listening

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Cocktail-party listening (CPL) refers to everyday listening situations where sounds from multiple sources compete for our attention. Normal-hearing individuals differ widely in their ability to perform in CPL experiments and such differences have been shown to be largely related to behavioral measures of the effects of internal noise. The source of this internal noise remains unknown, but some evidence exists to suggest that body noise (breathing, blood flow, muscular movements etc.) is a contributing component. Mostly notably, Steipan et al. (2019) have recently reported a significant correlation between noise levels measured in the ear canal and overall performance on the QuickSIN test (a CPL-like task). The goal of the present study was to provide a stronger test of this hypothesis by (1) measuring noise levels in the ear canal in a CPL task and (2) correlating these measures to noise efficiency – a direct measure of the behavioral effect of internal noise. Results fail to support a strong contribution of body noise to individual differences in the CPL task [work supported by NIDCD R01-DC001262].

Typical speech exposure during fetal auditory neurodevelopment

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There are many factors that influence language acquisition in young children, potentially affecting their language abilities later in life. Although it has been shown that fetal auditory exposures during the third trimester can shape neural responses to speech and language for newborns, we have yet to determine if and how prenatal auditory experience plays a role in shaping later language development. The fetal auditory system begins to respond to extrauterine sounds as early as 23 weeks' gestation. To begin examination of speech exposure during this period, we had pregnant women wear small 24-hr recording devices during their second and third trimesters of pregnancy. We collected over 6700 hours of data and analyzed the variation in speech exposure during typical fetal development. Overall, fetuses were exposed to an average of

approximately 4.5 hours of speech per day. Variability between subjects was high, with daily averages ranging between 3 and 6 hours per day. This finding suggests that some infants are born with 50% less extrauterine speech exposure than their peers. Whether this variability in speech exposure during fetal development affects language skills and abilities later in childhood remains an open question.

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Access to phonetic information at extended high frequencies improves speech-in-speech performance

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Recent studies from our lab and others have investigated the utility of the extended high frequencies (EHF, defined as frequencies ≥ 8 kHz) for speech perception. Many studies examining speech-in-speech listening (*i.e.*, the cocktail party problem) simulate an unnatural scenario where the target talker and maskers are all facing the listener. We analyzed listeners' performance in a more realistic listening scenario with a target talker facing the listener, and co-located maskers having head orientations facing away from the listener (45 or 60 degrees relative to the listener). By comparing full-band and bandlimited conditions (low-pass filtering at 8 kHz), we found that access to EHF information under our more ecological listening conditions provided an advantage to listeners. We questioned whether temporal or phonetic information extracted from EHF provided this advantage. Thus, we assessed performance when listeners were only given EHF temporal cues in the full-band condition. Results indicate that access to temporal cues alone are sufficient to improve speech-in-speech performance, but access to phonetic information provides additional gains. These results indicate that phonetic information is present in the EHF and is utilized by young, normal-hearing listeners.

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Singing Proficiency in children with bimodal hearing and bilateral cochlear implants

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The purpose of the present study was to investigate the pitch accuracy of vocal singing in children with bimodal hearing or bilateral cochlear implantation in comparison to age-matched normal-hearing (NH) children. The participants included 34 children with NH, 13 children with bimodal hearing [cochlear implant (CI) in one ear and hearing aid (HA) in the other ear], 31 children with sequential bilateral CIs, and 10 with simultaneous bilateral CIs. All participants aged between 7 and 12 years old. Each participant was recorded singing self-chosen songs that were most familiar to him or her. The fundamental frequencies (F0s) of individual sung notes were extracted. Four pitch-based metrics including contour direction, pitch variance ratio of entire song, mean deviation of F0 across notes, and mean deviation of the pitch intervals were computed and used as measures of vocal singing proficiency. The results revealed significantly poorer performance on all four pitch-based metrics in the three groups of pediatric CI recipients in comparison to the NH controls. No significant difference was found between the children with bimodal devices and bilateral CIs. Principal component analysis and stepwise regression will be performed to examine the potential contributing factors such as age at CI, age at HA, duration of HA use, etc. for the highly variable performance in singing among the hearing-impaired children.