

# Model simulations of masked thresholds for tones in dichotic noise maskers (A)

**Citation for published version (APA):**

Holube, I., Colburn, H. S., Par, van de, S. L. J. D. E., & Kohlrausch, A. G. (1995). Model simulations of masked thresholds for tones in dichotic noise maskers (A). *Journal of the Acoustical Society of America*, 97(5), 3411-3412. <https://doi.org/10.1121/1.412493>

**DOI:**

[10.1121/1.412493](https://doi.org/10.1121/1.412493)

**Document status and date:**

Published: 01/01/1995

**Document Version:**

Publisher's PDF, also known as Version of Record (includes final page, issue and volume numbers)

**Please check the document version of this publication:**

- A submitted manuscript is the version of the article upon submission and before peer-review. There can be important differences between the submitted version and the official published version of record. People interested in the research are advised to contact the author for the final version of the publication, or visit the DOI to the publisher's website.
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levitation. During the experiments, a liquid is poured on the surface of the acoustic transducer and is ultrasonically atomized. By applying a static electric field on the surface of the liquid, the atomized drops are electrically charged. After the drops are dispersed into the levitator, they can be contained together and levitated by acoustic radiation force. Coalescence among the drops is prevented by the forces of electric repulsion. The separation distance between the drops, the droplet size, and the number of levitated drops can be controlled by the intensity of the acoustic and electric fields. Preliminary experimental results will be reported, including the observation of mass transfer among the levitated drops. [Work supported by NASA through JPL, contract 958722.]

4:15

**5pPAb7. Nonlinear acoustics of bubbly liquids: Traveling waves in a quadratic approximation.** Daniel Goldman, Ali Nadim, and Paul E. Barbone (Dept. of Aersp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

An important aspect of underwater acoustics is the effect of trapped clouds of air bubbles. Recently, an approach to wave propagation in dilute bubbly liquids was described in which the Rayleigh-Plesset equation is used to derive a nonlinear equation of state for the gas bubble-liquid mixture [Nadim *et al.*, Bull. Am. Phys. Soc. 39, 1976 (1994)]. In this formulation, the pressure in the mixture depends in a highly nonlinear way on the local density and its first and second time derivatives. Linearization leads to the standard acoustic equation for bubbly liquids. Here the quadratic approximation to the nonlinear equation of state is examined, along

with the related nonlinear wave equation. The high- and low-frequency limits are then analyzed, and traveling wave solutions are sought in each case. Comparison is made with existing results for nonlinear waves in bubbly media.

4:30

**5pPAb8. Feasibility of low-frequency single-bubble sonoluminescence.** Robert E. Apfel, Tao Shi, Joseph Jankovsky, Jeffrey Ketterling, and Xiaohui Chen (Dept. of Mech. Eng., Yale Univ., New Haven, CT 06520-8286)

The potential to perform single-bubble sonoluminescence (SBSL) at low frequencies is motivated by the payoff of greatly enhanced energy concentration during collapse. Yet it is also known that bubbles undergoing such catastrophic collapse tend to be unstable. Experimental apparatus has been designed and computer simulations have been performed to test the feasibility of low-frequency, single-bubble sonoluminescence. The experimental apparatus consists of a cylindrical cell that is driven by an aluminum, half-wavelength resonator with fundamental resonance of less than 15 kHz. The cell is designed to be pressurized up to 5 atmospheres to allow levitation without significant spurious cavitation in the liquid. To complement this experimental work, our computer simulations of this phenomena are continuing [T. Shi and R. Apfel, J. Acoust. Soc. Am. 96, 3253 (1994)] in order to follow the shape distortion of collapsing bubbles for varying parameters, including acoustic frequency. The basic characteristics of our low-frequency resonator and the results of experimental work and computer simulations will be reported.

SATURDAY AFTERNOON, 3 JUNE 1995

AUDITORIUM, 1:15 TO 5:45 P.M.

### Session 5pPP

## Psychological and Physiological Acoustics: Binaural and Spatial Hearing; Cochlear and Auditory Nerve Function

Kim S. Abouchakra, Chair

U.S. Army Research Laboratory, AMSRL-SD-HR, Building 520, Room 39, Aberdeen Proving Ground, Maryland 21005-5425

### Contributed Papers

1:15

**5pPP1. A dynamic, psychophysical model of adaptation in localization experiments.** Barbara Shinn-Cunningham (Res. Lab. of Electron., MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Previous psychophysical models have described how resolution depends upon the range of physical stimuli employed in a given experiment; however, these models generally assume that subject performance is stable over time. The current work extends the context-coding model of Durlach and Braida [J. Acoust. Soc. Am. 46, 372-383 (1969); Braida and Durlach, J. Acoust. Soc. Am. 51, 483-502 (1970)] to account for changes in subject performance that occur when feedback is used to retrain subjects during the course of an experimental session. In the current model, observed changes in performance are accounted for by assuming a single exponential adaptation process. This process, which describes the adaptive state of the subject, determines the decision criteria and context (effective stimulus range) used in the model at a given point in time. From the dynamic decision criteria and effective range, and knowledge of the underlying sensitivity to localization cues, the model predicts mean response, bias, and resolution. The model predictions are in good agreement with results from a relatively large body of experiments, including experiments in which the number and range of stimuli were varied.

1:30

**5pPP2. Model simulations of masked thresholds for tones in dichotic noise maskers.** Inga Holube, H. Steven Colburn (Dept. of Biomed. Eng., Boston Univ., 44 Cummington St., Boston, MA 02215), Steven van de Par, and Armin Kohlrausch (Inst. for Perception Res. (IPO), Eindhoven, The Netherlands)

The study of masked thresholds in dichotic noise maskers is important for understanding the processing in binaural hearing. To simulate these thresholds a psychoacoustically motivated perception model was used [T. Dau *et al.* (1995). "A quantitative model of the "effective" signal processing in the auditory system: I. Model structure," submitted to J. Acoust. Soc. Am.]. This model, which has been successfully applied to several monaural psychoacoustical experiments, was extended by an additional binaural processing unit. It consists of a filterbank, half-wave rectifier, low-pass filter, and adaptation loops, which model the temporal processing. The binaural processing unit detects the interaural correlation and makes decisions based on the difference between the signals from both ears. Masked thresholds in the NoS $\pi$  and N $\pi$ So configurations, obtained as a function of noise masker frequency and bandwidth, were simulated and compared to new experimental measurements. The dependence on interaural delay and interaural decorrelation of the noise masker was also modeled and compared to data in the literature. In general, model simula-

tions agree well with the main features seen in the measurements. [Work supported by DFG (Ho 1627/1-1) and by NIDCD (Grant DC00100).]

1:45

**5pPP3. The index of interaural envelope correlation: Normalized cross-covariance or normalized cross correlation?** Leslie R. Bernstein and Constantine Trahiotis (Ctr. for Neurological Sci. and Surgical Res. Ctr., Dept. of Surgery, Univ. of Connecticut Health Ctr., Farmington, CT 06030)

This study principally evaluated whether the normalized cross covariance (Pearson product-moment correlation) or the normalized cross correlation describes discriminability of changes in interaural disparities conveyed by the stimulus envelope. In a four-interval, two alternative task, listeners detected which interval contained a 4-kHz tone added antiphasically to diotic, 200-Hz-wide, noise (NoS $\pi$ ). The "nonsignal" intervals contained the tone added homophasically (NoSo). Discriminability ( $d'$ ) was measured as a function of S/N for values between -30 and +30 dB (really!). For all S/N's, overall level was 70 dB SPL. Listeners' performance was very well accounted for by the normalized cross correlation but not the normalized cross covariance. Additionally, listeners were tested in a "direct" discrimination task where changes in envelope correlations ( $\Delta\rho$ ) were produced by "mixing" two independent Gaussian noises. Although  $\Delta\rho$ 's at threshold ( $d'=1.0$ ) obtained from the two tasks were similar, the psychometric functions obtained with the direct discrimination task were more steep. Discussion will include how the normalized cross correlation of the envelope accounts for classic data concerning discriminability of interaural time differences at high frequencies as a function of depth of modulation. [Work supported by NIH DC02103.]

2:00

**5pPP4. Precedence and plausibility.** William A. Yost and Sandra J. Guzman (Parmlly Hear. Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

The "Clifton effect" [R. K. Clifton, J. Acoust. Soc. Am. **82**, 1834-1835 (1987)] was studied in a sound-deadened room with seven loudspeakers. One loud speaker produced a source click while other loudspeakers produced delayed copies simulating echoes (delays: 2-30 ms). Each combination of source and echoes is one click event and was presented as a train of click events (1-20 click events). A train was presented to listeners who made two judgments for the LAST click event presented: (1) The number of loudspeakers which produced sounds for the last click event, and (2) the loudspeaker location for each perceived source. "Catch trials" were introduced to make sure listeners used all possible responses and were able to locate the loudspeaker sources. When more than 10 click events were presented, a switch in conditions was introduced between the 10th and 11th click event. If the switch was plausible for a natural source and its echoes, responses indicated that listeners processed delayed clicks as echoes. If the change was implausible, then responses after the switch changed indicating listeners processed all clicks as if they were sources rather than echoes. [Work supported by NIH and AFOSR.]

2:15

**5pPP5. Lateralization of high-frequency FM tones and frequency sweeps.** Kourosh Saberi (Psychoacoust. Lab., Dept. of Psychol., Univ. of Florida, Gainesville, FL 32611)

Lateralization thresholds were measured in three experiments. In experiment I, thresholds measured for sinusoidal FM with carriers of 3 or 4 kHz and modulation rates from 50 to 800 Hz were comparable to those measured for sinusoidal AM stimuli. Lowest thresholds (at 71% correct) were about 100  $\mu$ s, obtained at modulation rates of 300-350 Hz. In experiment II, a 300-ms tone was linearly swept in frequency from 2 to 5 kHz. The slope and intercept of the time-frequency function were randomized by 10% on each observation. Lateralization thresholds were about 50  $\mu$ s. Unequal time-frequency slopes at the two ears produced a sense of motion. In experiment III, a sinusoidal FM was presented to one ear and a sinusoidal AM to the other ear. When the FM and AM had the same

modulation rate (250 Hz), a single image was perceived. Observers were sensitive to the interaural modulation phase with near perfect discrimination of homophasic from antiphasic conditions. Results are discussed in terms of an FM-to-AM transduction mechanism. [Work supported by NIH and AFOSR.]

2:30

**5pPP6. The segregation of SAM 4-kHz targets from SAM 2-kHz distractors on the basis of interaural envelope delay.** R. H. Dye (Parmlly Hear. Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

A stimulus-classification paradigm was used to examine the extent to which judgments of the laterality of 4-kHz targets, sinusoidally amplitude modulated at 200 Hz, were influenced by 2-kHz distractors that were modulated at rates ranging from 50 to 400 Hz. On each trial, the target was presented with one of ten envelope delays (ranging from -250 to +250  $\mu$ s in 50- $\mu$ s steps), as was the distractor. Each test interval was preceded by a diotic presentation of the target alone. The duration of the signals was 200 ms. During a block of 100 trials, each combination of target-distractor delay was presented once. The relative salience of the envelope delays carried by the target and the distractor was assessed by the slope of the best linear boundary between left and right responses. Two listeners gave increasing weight to the target as the difference between the target and distractor modulation frequencies increased, but weighed that target and distractor equally when both were modulated at 200 Hz. Two other listeners gave increasing weight to the 2-kHz distractor as modulation frequency increased, as though its relative salience increased with "number of looks." These data will be compared to measures of binaural interference that have been obtained for similar stimuli. [Work supported by NIH.]

2:45

**5pPP7. Precision of sound localization measured by a reaching task.** Daniel H. Ashmead, Xuefeng Yang, Robert Wall, and Kiara Ebinger (Dept. of Hear. & Speech Sci., Vanderbilt Univ. Med. Ctr., Nashville, TN 37232-8700)

This study validates a reaching measure of sound localization for subsequent application to children. Seven adults reached for broadband sound sources while hand position was measured to within 2 mm. Sounds came from 18 regions in frontal reaching space, with the loudspeaker moved away just after stimulus offset. In the "visual" condition subjects watched until the sound ended, then closed their eyes and reached. In the "auditory" condition subjects were blindfolded. Precision of sound localization was estimated by comparing variability in the visual and auditory conditions:  $s = \sqrt{s_A^2 - s_V^2}$ . Estimates were computed for horizontal angle, vertical angle, and distance for each target location. Results agreed reasonably well with conventional measures. For targets straight ahead at ear level, horizontal  $s=2.6^\circ$ , vertical  $s=5.0^\circ$ , and distance  $s=9.5\%$ . Systematic variations occurred across target locations. This reaching task is a rapid, naturalistic way of measuring three-dimensional sound localization. [Work supported by DOE and NIH.]

3:00

**5pPP8. Virtual auditory reality reduced to the bare essentials.** William Morris Hartmann and Andrew Wittenberg (Michigan State Univ., Dept. of Phys., East Lansing, MI 48824)

Successful imaging of real sound sources by headphones can be done by measuring free-field head-related transfer functions (HRTF) using small microphones in the ear canals and then inverse filtering by the headphone response. A simpler alternative to this standard procedure is described where a listener, with small microphones in the ear canals, wears open-air headphones throughout the experiment. A synthesized vowel is played, first from a loudspeaker and then from the headphones. The headphone signal is adjusted so that the amplitudes and phases of the harmonics measured with the small microphones are the same as those found when the loudspeaker is sounding. The technique is successful in that listeners cannot distinguish between real and virtual sources. It lacks the flexibility