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<p>The attached report covers the second year of the project. Testing of the revised model of auditory processing developed in the first year was initiated at the University of Kansas. The PI moved from Kansas in August 1988, to The Ohio State University. The remainder of the work for the year was conducted under a sub-contract from Kansas to the Ohio State University Research Foundation. Testing of normal listeners in the frequency glide vs multiple-step transition task has indicated that the normal ear has a temporal window of approximately 7 to 10 msec. Further, these results appear to indicate that the critical band, thought to be ubiquitous in peripheral processing, has no effect on the listeners' discriminations of sub-critical, critical or supra-critical bandwidth swept frequency signals.</p>			
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DEMODULATION PROCESSES IN AUDITORY PERCEPTION

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1 March 1989

Final/Annual Report for Period 1 December 1987 - 30 November 1988

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report for KN/FETH
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Bolling Air Force Base
Washington, DC 20332



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Dec. 1, 1987 - Nov. 30, 1988

Introduction

→ The overall goal of this project is to understand the ability of the human listener to extract information from complex, time-varying sounds such as speech, music or other environmentally important signals. Specifically, we are interested in the listener's ability to process modulations of frequency and amplitude which are thought to carry the information in such signals. To that end we have devised a signal-processing model that calculates the Envelope-Weighted Average of the Instantaneous Frequency (EWAIF) for complex, time-varying signals. Previous research had shown that the listener's ability to distinguish one complex sound from another can be predicted by calculating the EWAIF value for each sound.

The original version of the EWAIF model processed an entire signal, regardless of its duration, as a whole. In the first year of funding for this project, we revised the model to incorporate a temporal processing window. The revised model acts as a short-term running averager: A brief segment of the signal is "captured" within the window. The EWAIF calculation is performed on the windowed segment, before the window is move in time to capture a new segment. Our original version of the running-average EWAIF model incorporated two parameters to describe the windowing process. Both the width of the window and the lag (i.e., the temporal overlap) were specified. Recent efforts to model listener performance have used a moving average "filter" approach, thus only one parameter is necessary.

→ We initiated a series of experiments to test the performance of the new EWAIF model. Listeners were asked to discriminate between two frequency modulated tones. For one tone, the GLIDE, the frequency began at a given value and changed linearly over time to a final value. The extent of the transition, in Hertz, and its duration, in msec., were parameters of the experiment. For the target signal in the two-alternative, forced-choice task, the tone's frequency began at the same frequency and moved to the same final frequency over the same time period; however, the trajectory was different. The target signal followed a series of discrete steps in moving from the initial to final frequency. For a small number of discrete steps in this STEP signal, the listener can easily distinguish the GLIDE from the STEP. As the number of steps is increased, and thus, the extent and duration of each individual step is reduced, listeners have more difficulty in making the required discrimination.

The discrimination of STEP vs. GLIDE signals was begun in spring semester 1988 at the University of Kansas. In August 1988, the PI moved to the Speech and Hearing Science Division at the Ohio State University. The remaining four months of work was performed under a research sub-contract from the University of Kansas to the Ohio State University Research Foundation. Data collection was interrupted for the months of August and September 1988, while the laboratory equipment was being moved from Lawrence, KS to Columbus OH, reassembled and tested. Further delay was encountered because Ohio State is on a quarter system and classes did not begin until late in September. We were unable to recruit a new group of listeners until the students returned for the fall quarter.

Data collection begun in the fall quarter, continued through the end of the period covered by this report (Nov. 30, 1988) into the current period funded directly through Ohio State. A competing renewal of the project was submitted in August 1988. Work on the project has continued since Nov. 30, funded directly through Ohio State.

In addition to the experimental portion of the work, the implementation of the EWAIF model was revised and extended. Originally, we implemented the model within the ILS signal processing package on the PDP 11/23 laboratory computer at Kansas. At Ohio State we have moved upward and downward from the 11/23. The model is implemented on a SUN workstation system using the MATLAB package and also on a PC in Turbo Pascal. The implementation on the SUN incorporates the front-end of the Patterson Pulse Ribbon model of pitch perception which simulates the peripheral filtering of the auditory periphery. This implementation was accomplished with the considerable help of Ying Yong Qi, then a doctoral student in Speech and Hearing Science, and Ashok Krishnumurthy, Assistant Professor of Electrical Engineering at Ohio State. Both continue to work on the project in the current year.

List of research objectives and current status

1. Continue the STEP vs GLIDE discrimination testing at other signal frequencies.

Discrimination testing was begun for frequency excursions of 100-, 200- and 400 Hz, centered at 1 kHz and 2 kHz, with durations of 25-, 50- and 100 msec. Work was completed on four listeners at those frequencies at Kansas, before the move to Ohio State. At Ohio State three new listeners were recruited and the work at Kansas was partially replicated to ensure that the laboratory equipment was in fact not damaged by the relocation. The listeners at Ohio State were then tested at 500 Hz and 4 kHz. Agreement between results from Kansas and those from Ohio State is extremely good. In fact, agreement across subjects is remarkable good.

Results of the discrimination testing were plotted as percent correct discriminations as a function of the duration of an individual step. That is, if the STEP signal contained four steps over 100 msec, the P(C) score would be plotted at a step duration of 25 msec. P(C) for discrimination of a GLIDE vs 50-msec two-step signal would be plotted at the same 25 msec/step point. Psychometric functions were grouped according to the rate of change of the frequency. Thus, we have 8 Hz/msec, 4 Hz/msec and 2 Hz/msec rates for the conditions tested.

The psychometric functions, grouped as described above (see figures included with this report) reveal several very interesting effects. First, extent of transition in Hertz, or duration in msec, have little effect on performance when the psychometric functions are grouped by transition rate. Across the rates tested, performance differs little between 8 Hz/msec and 4 Hz/msec. At 2 Hz/msec the functions are slightly less steep. Performance is similar at 500 Hz, 1 kHz and 2 kHz for the listeners tested so far. At 4 kHz, all listeners show a marked decrement in discrimination performance. The task is much more difficult to accomplish. At the lower center frequencies, the 75% point estimate of a temporal window indicates a value of 7 to 10 msec. Estimates are more difficult to determine at 4 kHz, because of the listeners' inability to perform the task. Our preliminary explanation for the break down in performance at 4 kHz centers on the assumption that synchronization of nerve fibers must be necessary for the listener to track the frequency transitions.

Another surprising feature of these results is the lack of effect of the critical bandwidth. Except at 4kHz, the transitions sweep across more than one critical bandwidth, yet the listeners' performance is not different for sweeps within one critical bandwidth versus sweeps that traverse two or more critical bandwidths.

2. Implement a multiple-band version of the EWAIF model to handle wide bandwidth sounds.

Patterson's Pulse Ribbon Model for complex sound pitch perception was ported to a SUN/3 workstation system. We have attempted to insert the EWAIF calculation between the output of each "critical band" filter and its hair cell emulator. The model obtained from Patterson is running as expected for non-dynamic signals; however, the interaction of the narrow bandwidth filters with the swept-frequency signals used in our discrimination experiments has led to some unusual results. Figures included with this report indicate the problem. The instantaneous frequency of the filtered signal from each band shows deviations from the expected values. It appears as if the filter begins to respond at its natural frequency when the signal is first introduced. As the swept tone enters the pass band, the tone frequency takes over, but as the tone leaves the pass band, the filter response moves back toward the natural frequency. The nature of these deviations depends upon both signal and filter parameters. Our experimental results described above

reveal none of these interactions. Different implementations of the filter bank and a change from FFT to FIR filter implementations of the Hilbert transform have not altered this effect. These preliminary model results further support the notion that our swept tone signals are processed without the influence of the critical band.

3. Complete single-step vs glide discrimination experiments.

The series of single-step vs glide discrimination experiments was completed in summer 1988 by R. A. Gerren, In addition, listeners were tested on discrimination of Huffman sequence signals like those used by Patterson and Green in their temporal acuity study in 1970. A manuscript is being written by Gerren and Feth for submission in 1989.

Participating professionals

Lawrence L. Feth, Ph.D.	Principal Investigator
Lisa J. Stover, M.A.	Grad. Research Asst. (to 8/1/88)
Richard A. Gerren, Ph.D.	Postdoctoral fellow (to 8/1/88)
Mary E. Neill, B.A.	Grad. Research Asst. (fr 8/1/88)
Ying Yong Qi, M.S.	Grad. Research Asst. (fr 8/1/88)
Ashok Krishnumurthy, Ph.D.	Co-Investigator (fr 12/1/88)

Note R. A. Gerren was funded by an NIH institutional postdoctoral fellowship awarded through the University of Kansas. Y. Y. Qi was funded by an Ohio State pre-doctoral fellowship for the period covered by this report. As of March 1, 1988 he will be funded by a graduate school postdoctoral fellowship to continue until Nov 30, 1989. From Dec 1, 1989 on, he will be funded through this project.

Publications and Presentations

Complex sound discrimination: Predictions of the EWAIF model. L.L. Feth, L.J. Stover and R.A. Gerren. [Abstract: J. Acoust. Soc. Amer. 83, p. S35 (1988)] Presentation to the Acoustical Society meeting May 1988, Seattle WA.

Demodulation Processes in Auditory Perception, L.L. Feth, presentation to the Bioacoustics Laboratory Group at Wright-Patterson AFB Dec 4, 1988, and to the Central Ohio Chapter of the Acoustical Society of America, Feb 4, 1989.

Envelope-Weighted Average of Instantaneous Frequency Model for Auditory Processing of Complex Sounds, L.L. Feth, L.J. Stover, R.A. Gerren, and M. E. Neill. (Presented at the twelfth mid-winter meeting

of the Association for Research in Otolaryngology, Feb 1989.)

Patents and Inventions

No patentable inventions have resulted from this research.

Statements

In the previous annual report it was stated that a manuscript for publication was anticipated by spring/summer of 1988. Obviously, that manuscript has not been completed. The main reason for the lack of a finished manuscript was the PI's decision to move from the University of Kansas to Ohio State. In addition to the actual down time resulting from the move, the PI had to negotiate the release of grant related equipment from Kansas, write a sub-contract agreement for the continuation of this work at Ohio State, and write a three-year renewal proposal to support the work beyond Nov 30, 1988. From March 11 through Sept 7, 1989, the PI will be visiting the laboratory of Roy Patterson at the Applied Psychology Unit of the Medical Research Council in Cambridge, England. The purpose of the visit is to work on the combination of EWAIF model dynamic features with the multi-channel features of Patterson's pulse ribbon model of pitch perception. In addition, it should be possible to complete one or more manuscripts during the visit.

The move to Ohio State was made because of that university's renewed commitment to support research in speech and hearing science. Over the past year, Ohio State has committed funds to house the speech and hearing division in remodeled space in Pressey Hall on the Ohio State campus. Two full professors (the PI and one other) were hired as additions to the faculty. Further additions for next year include a speech scientist with a Ph.D. in physics to work on articulatory models of speech production. The university has also expended approximately \$500,000 to the purchase of research equipment for speech and hearing science. The commitment to support research in this area is apparent.

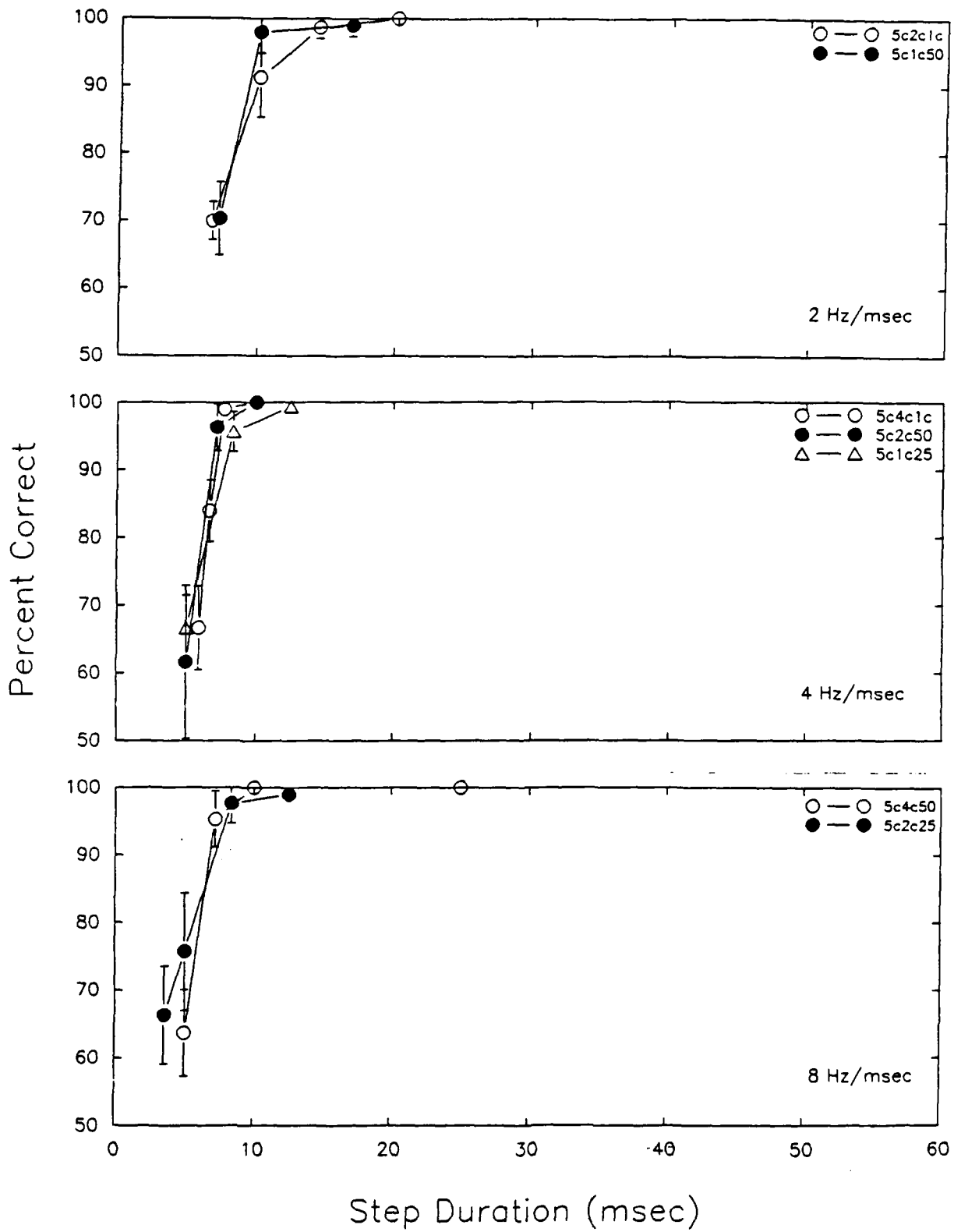
ENVELOPE-WEIGHTED AVERAGE OF INSTANTANEOUS FREQUENCY MODEL
FOR AUDITORY PROCESSING OF COMPLEX SOUNDS. *L.L. Feth¹, L.J. Stover²,
R.A. Gerren², M.E. Neill¹ ¹Speech and Hearing Science, Ohio State University,
Columbus, OH 43210, ²KASPL, University of Kansas, Lawrence, KS 66045.

The assumption underlying this work is that the human auditory system functions as an auditory signal processor. We consider the complex stream of sound to be a carrier modulated in amplitude and frequency. To extract the information borne by the carrier, the listener must be able to demodulate the sound stream. Much of the past work in psychoacoustics, and some of the present day effort, is based in what we call "spectrum picture processing". That is, the complex sound is Fourier analyzed into an amplitude-by-frequency picture in the experimenter's conception of the stimulus. Then, perception is modeled as if the spectral picture, not the complex, time-varying sound pressure wave, were the stimulus presented to the listener.

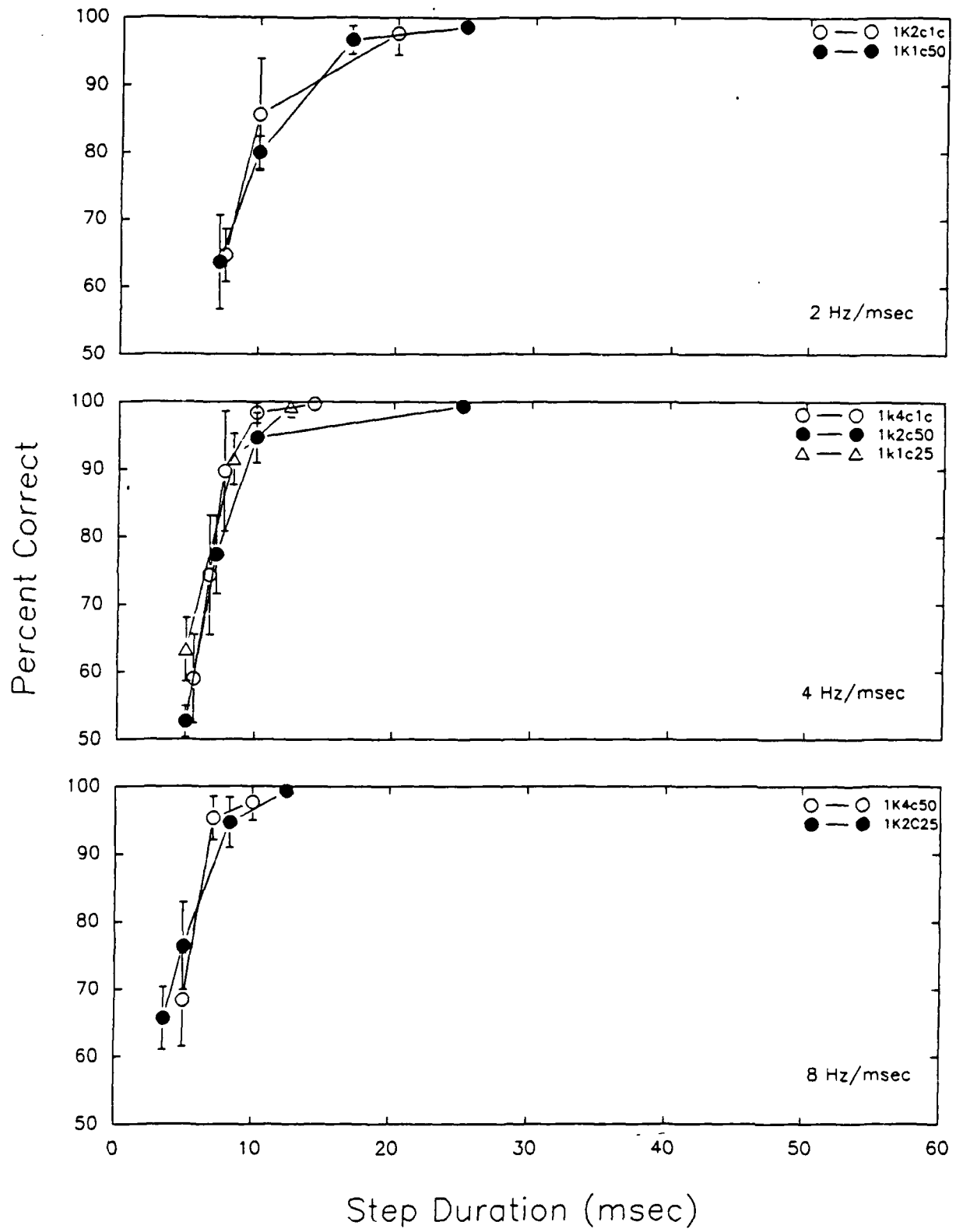
We have applied an earlier version of the EWAIF model to simultaneously amplitude- and frequency-modulated tones to predict the pitch that listeners perceive in such sounds. The model even predicts some confounding of listener performance for the signals used in the early profile analysis work. Our current implementation of the model tracks transitions in frequency and amplitude for narrow bandwidth signals. To test the model, we predict listener performance in distinguishing smooth linear transitions from multiple-step transitions over the same trajectory. For transitions centered from 250 Hz to 4000 Hz, at rates of 2-, 4-, and 8-Hz/sec, we find that listener performance can be characterized as a demodulation process. Limitations on the performance of our listeners will be compared with the predictions of the EWAIF model.

(Work supported by a grant from AFOSR)

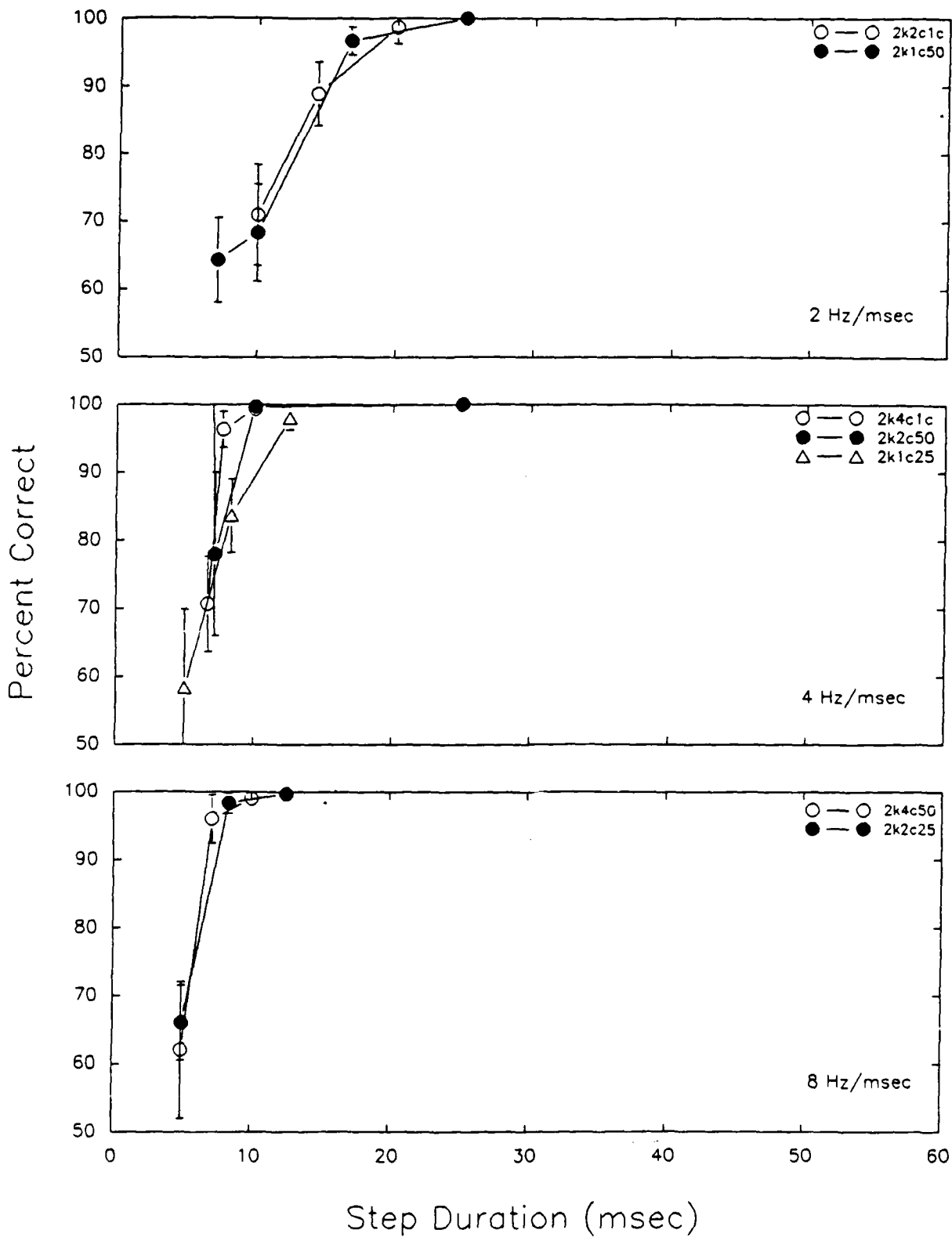
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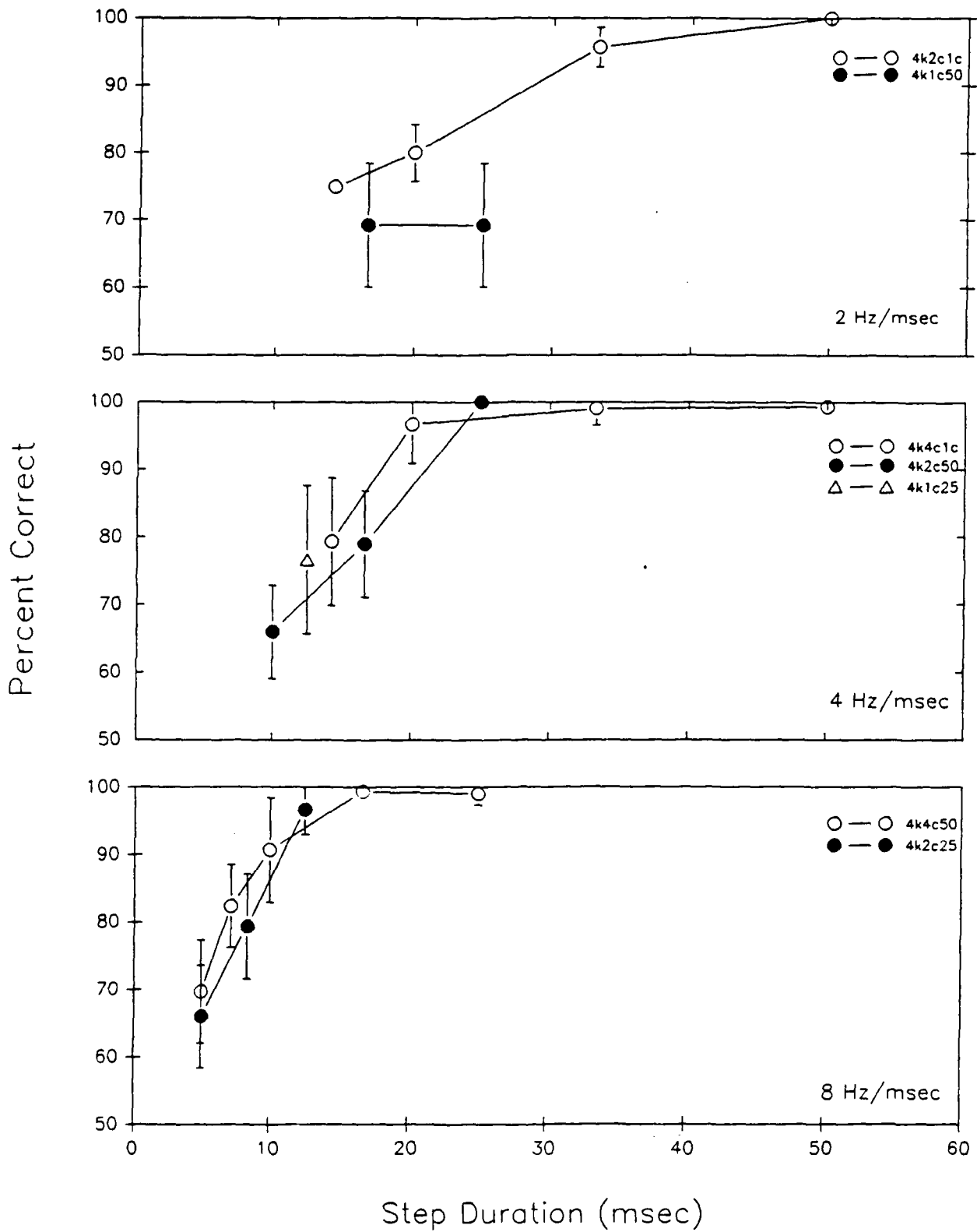
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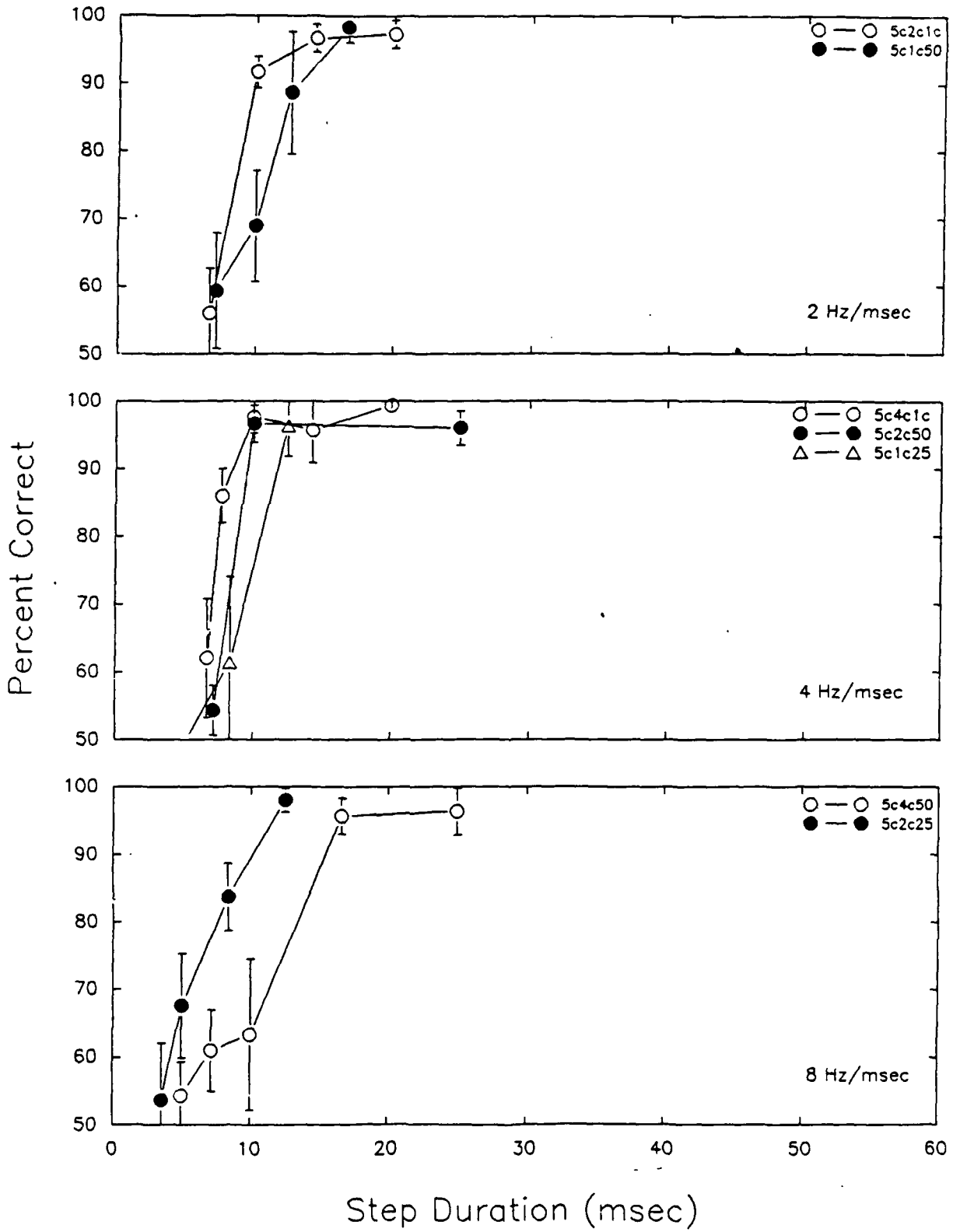
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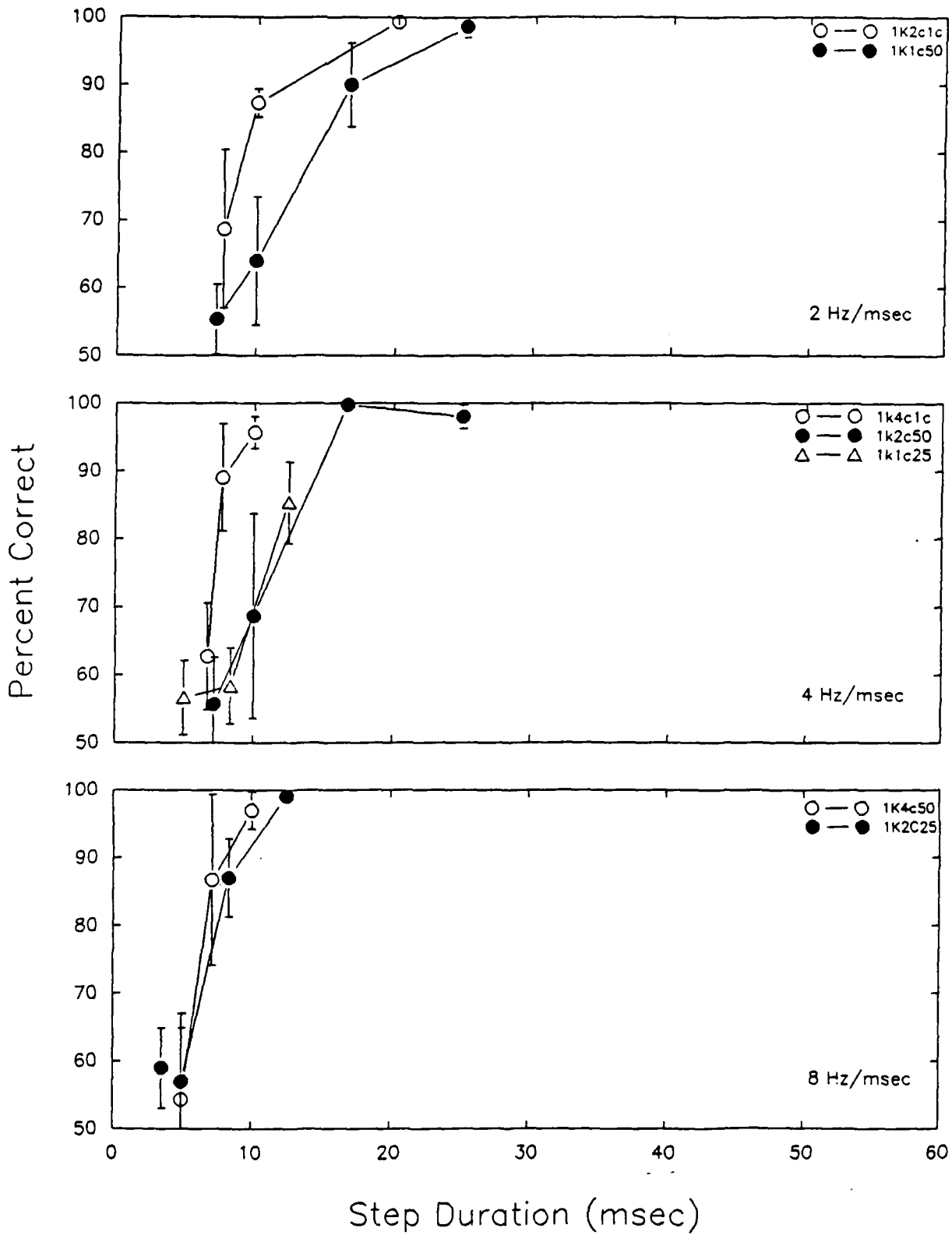
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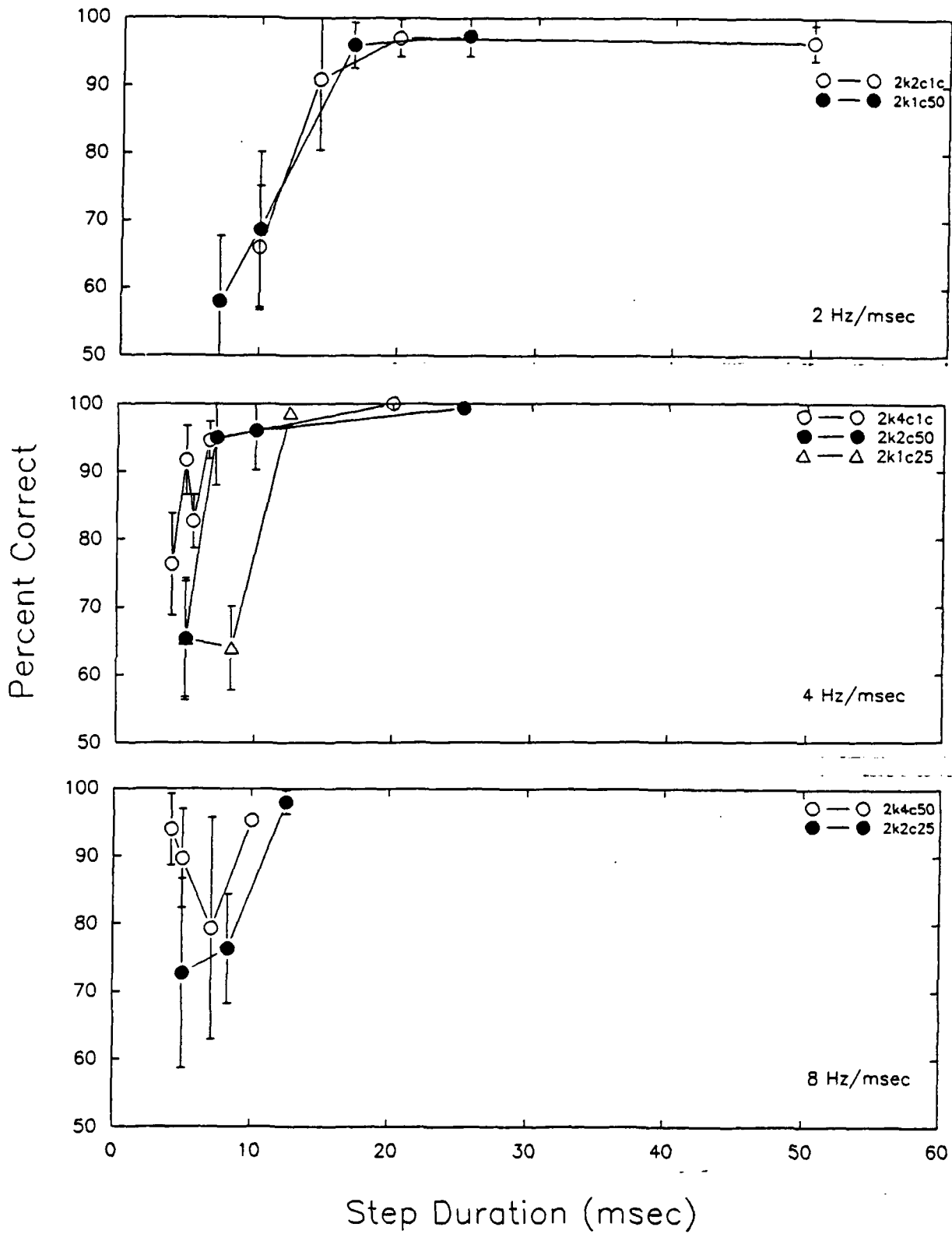
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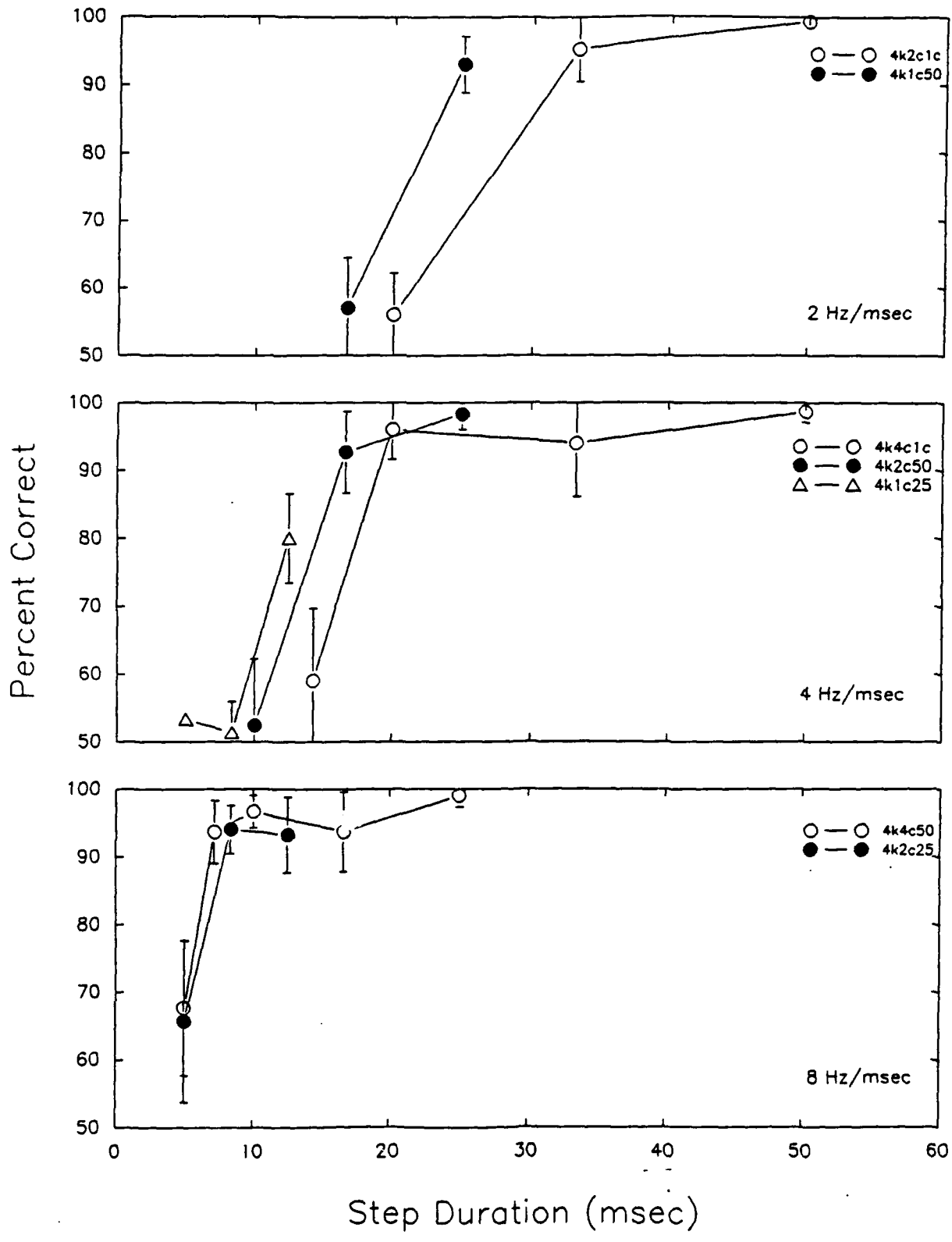
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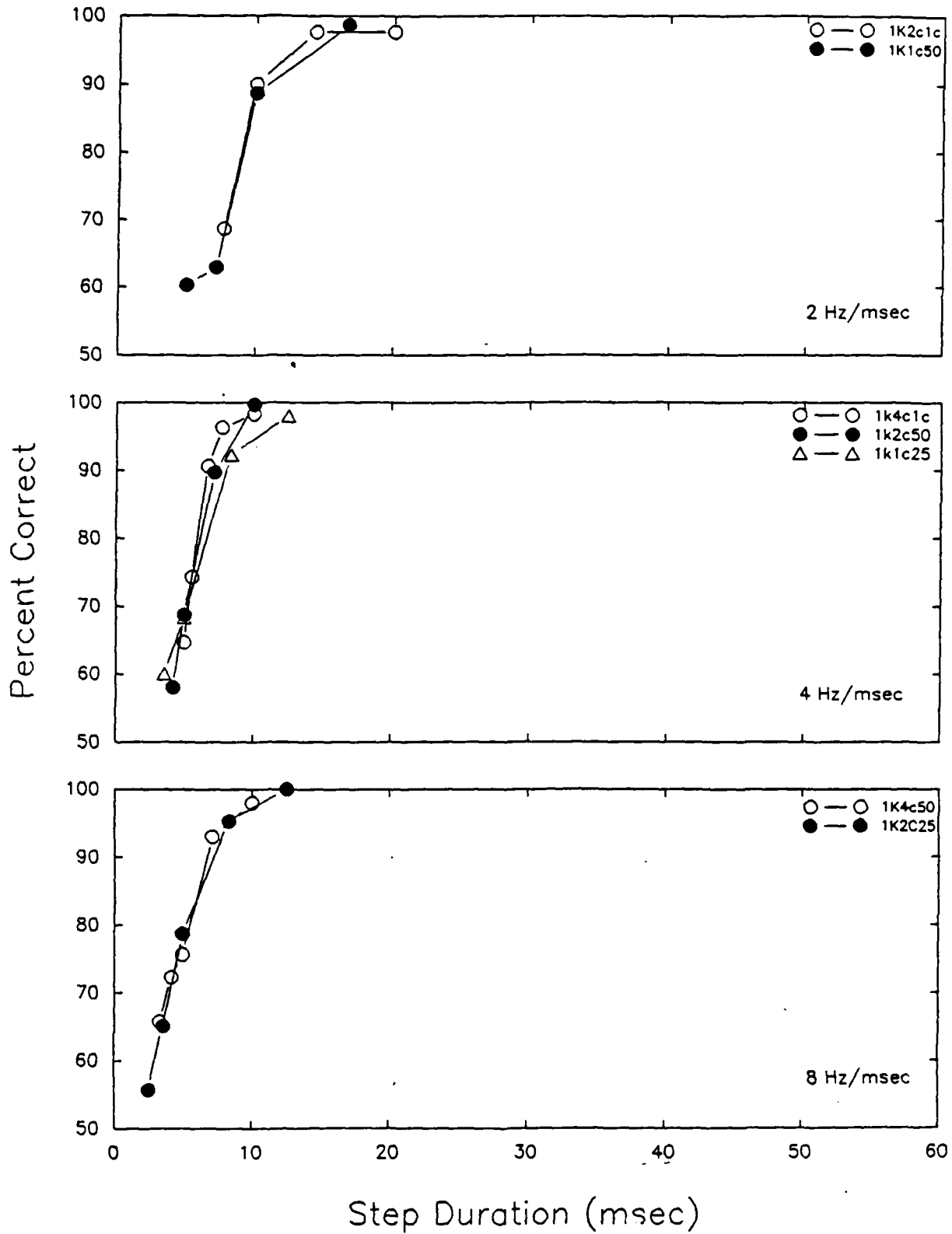
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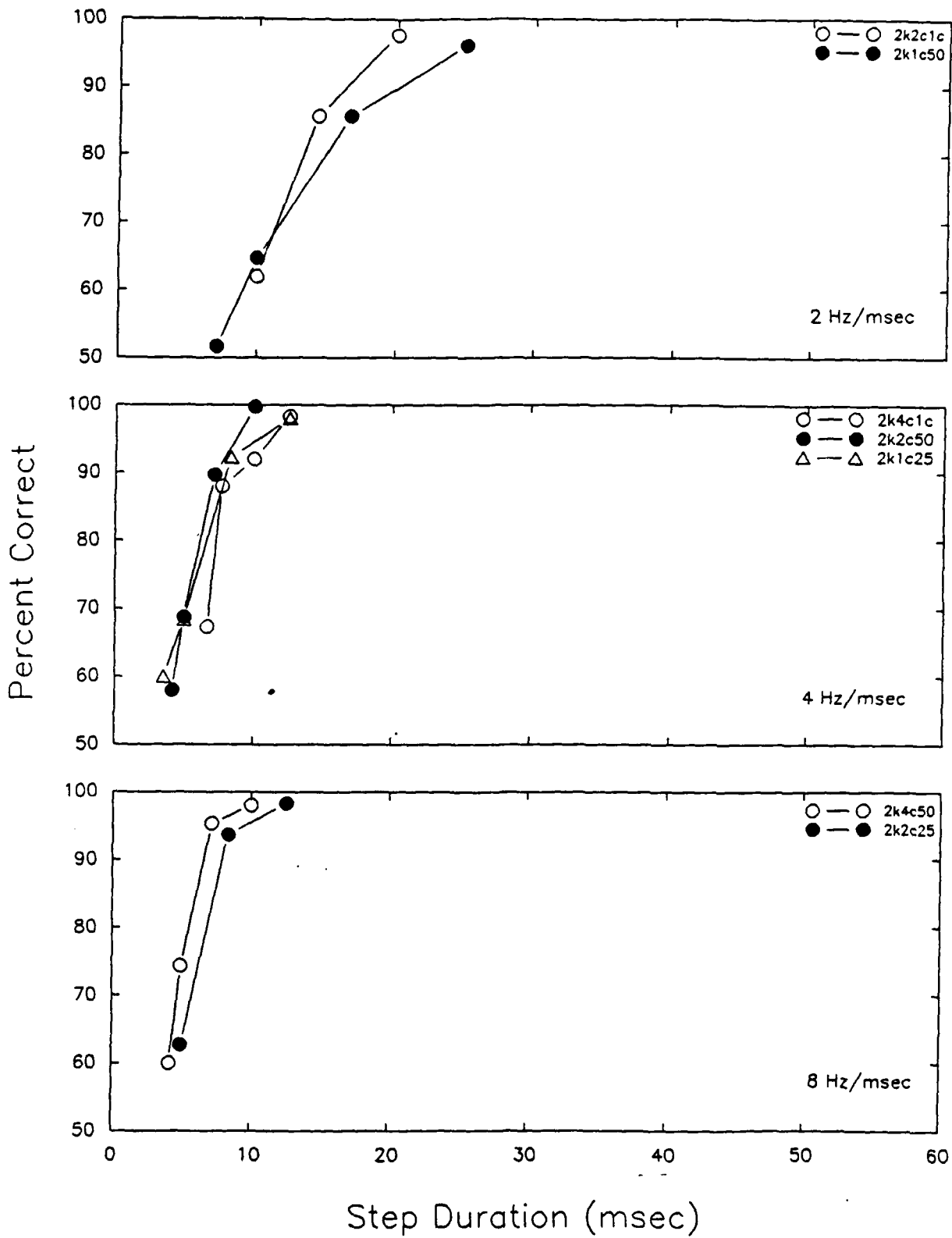
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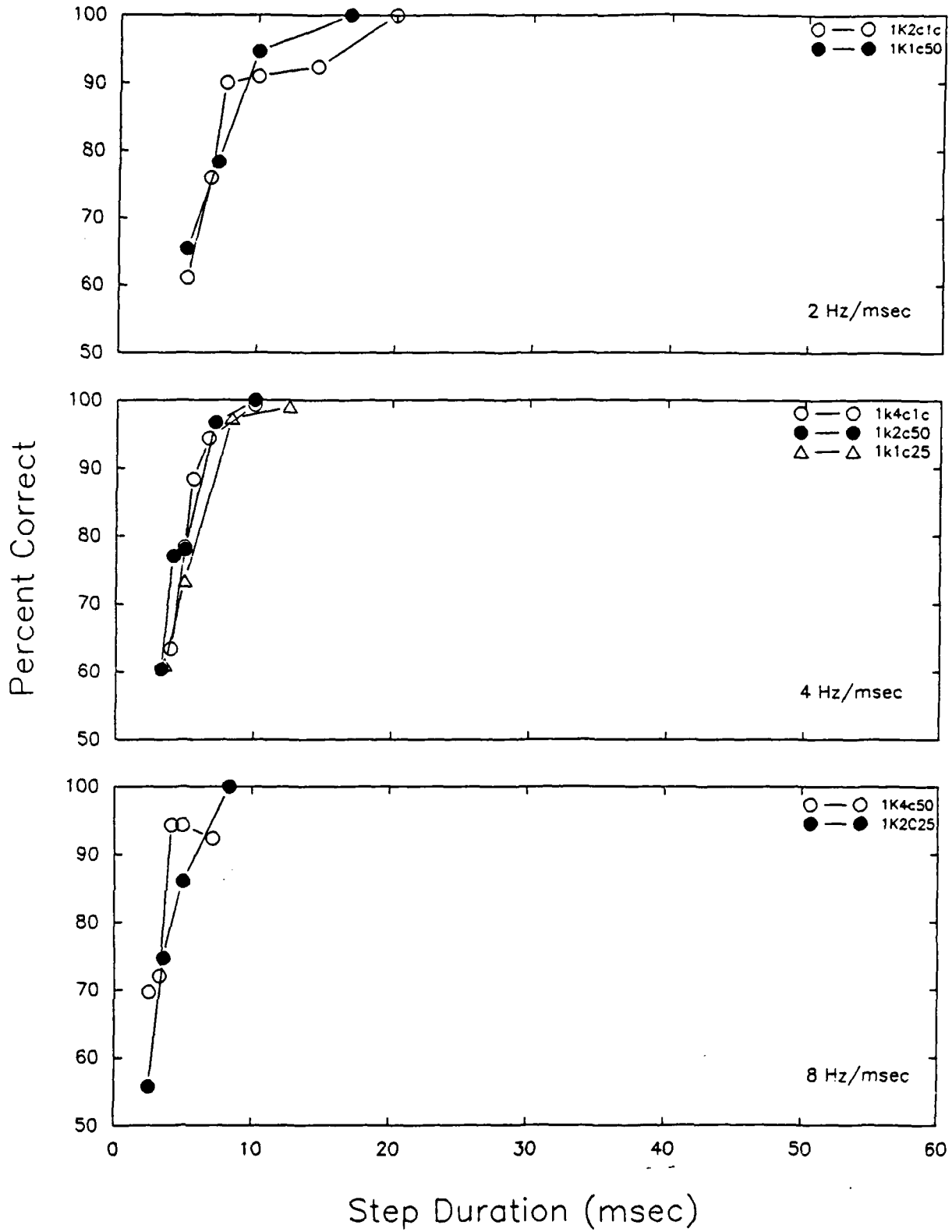
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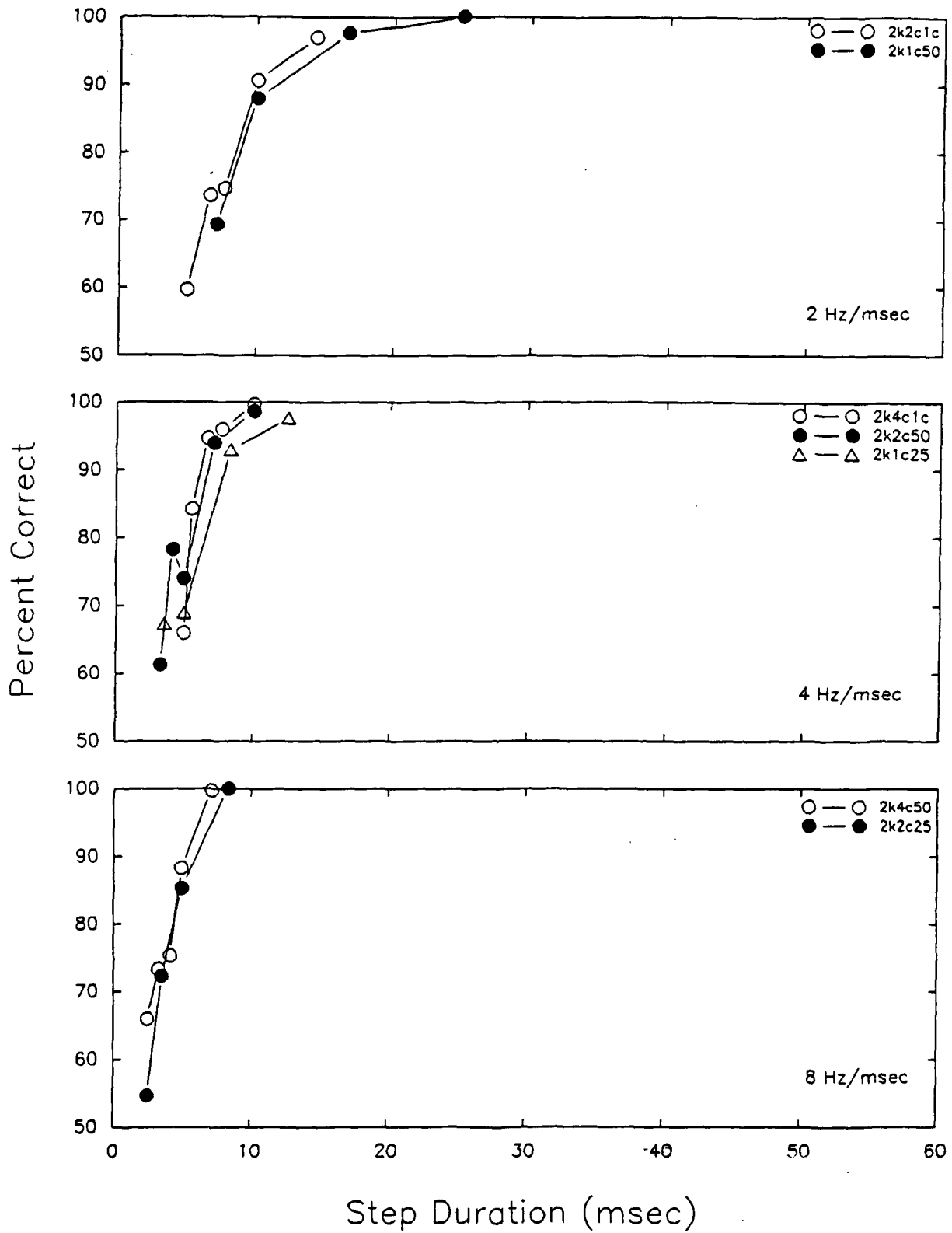
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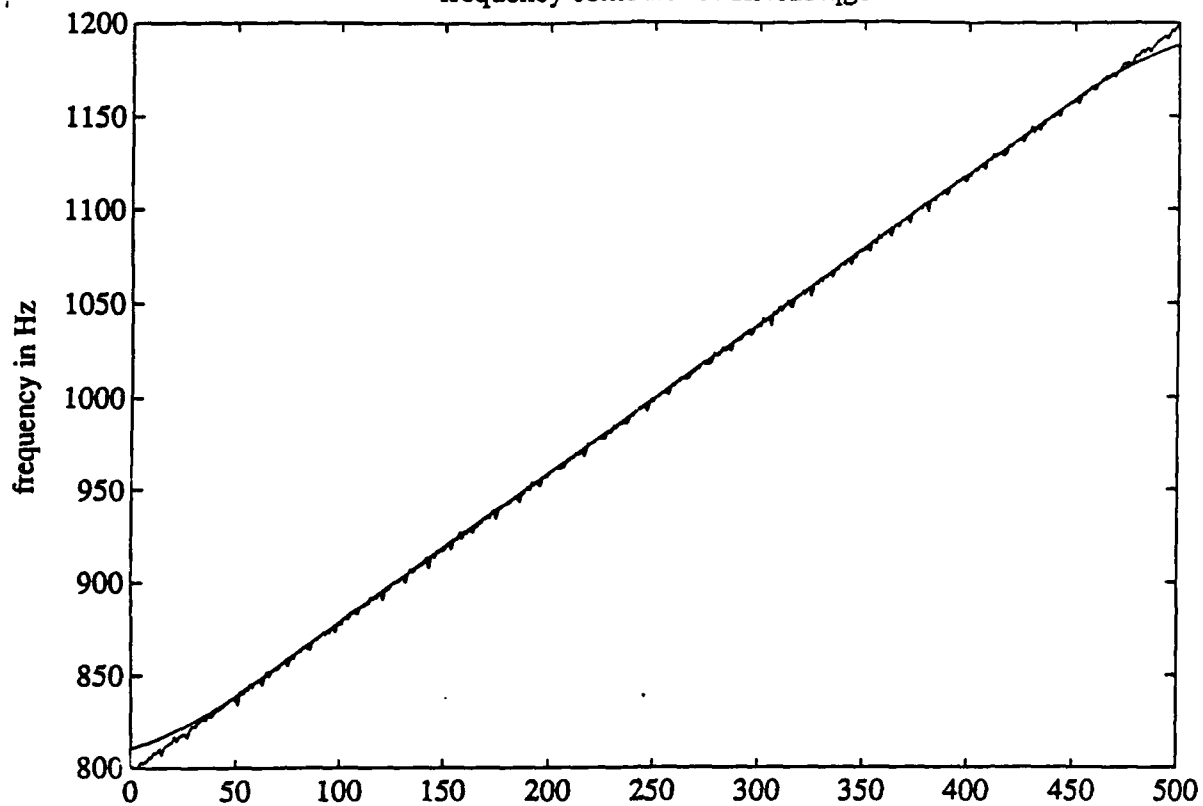
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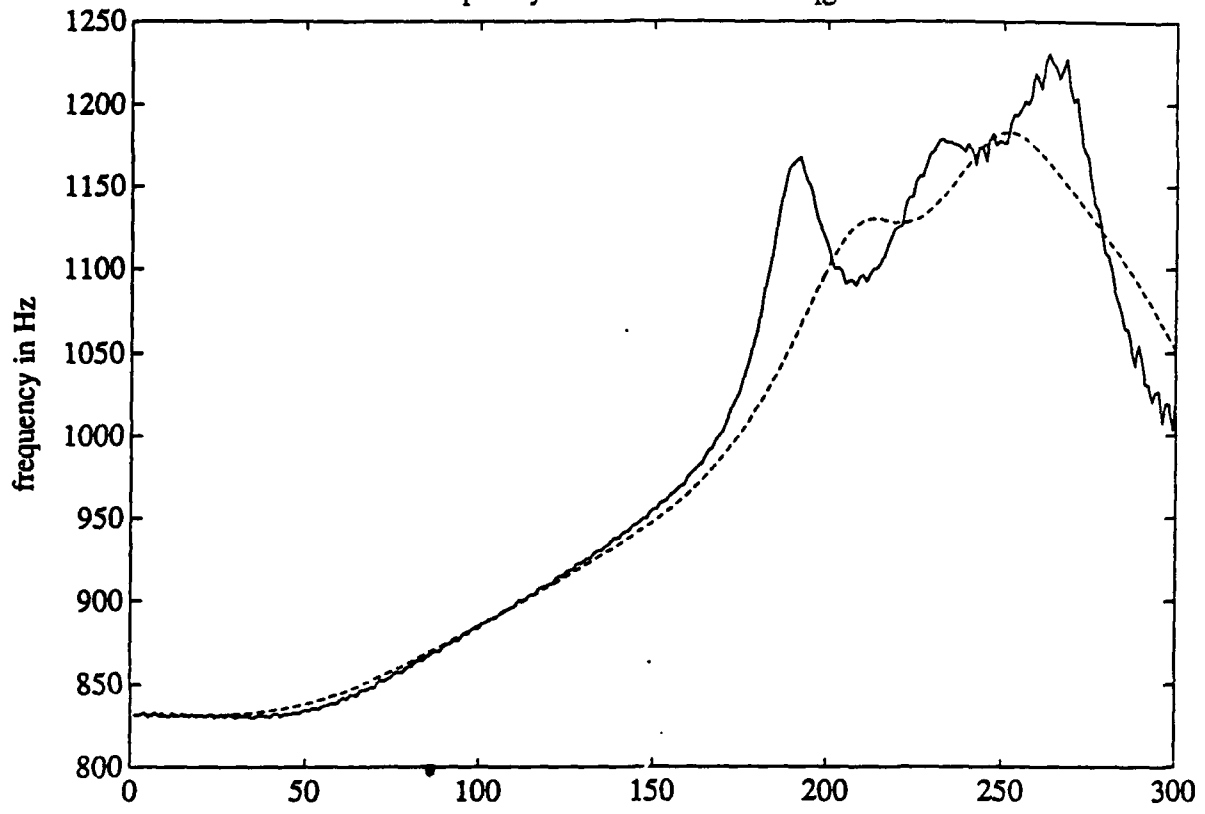
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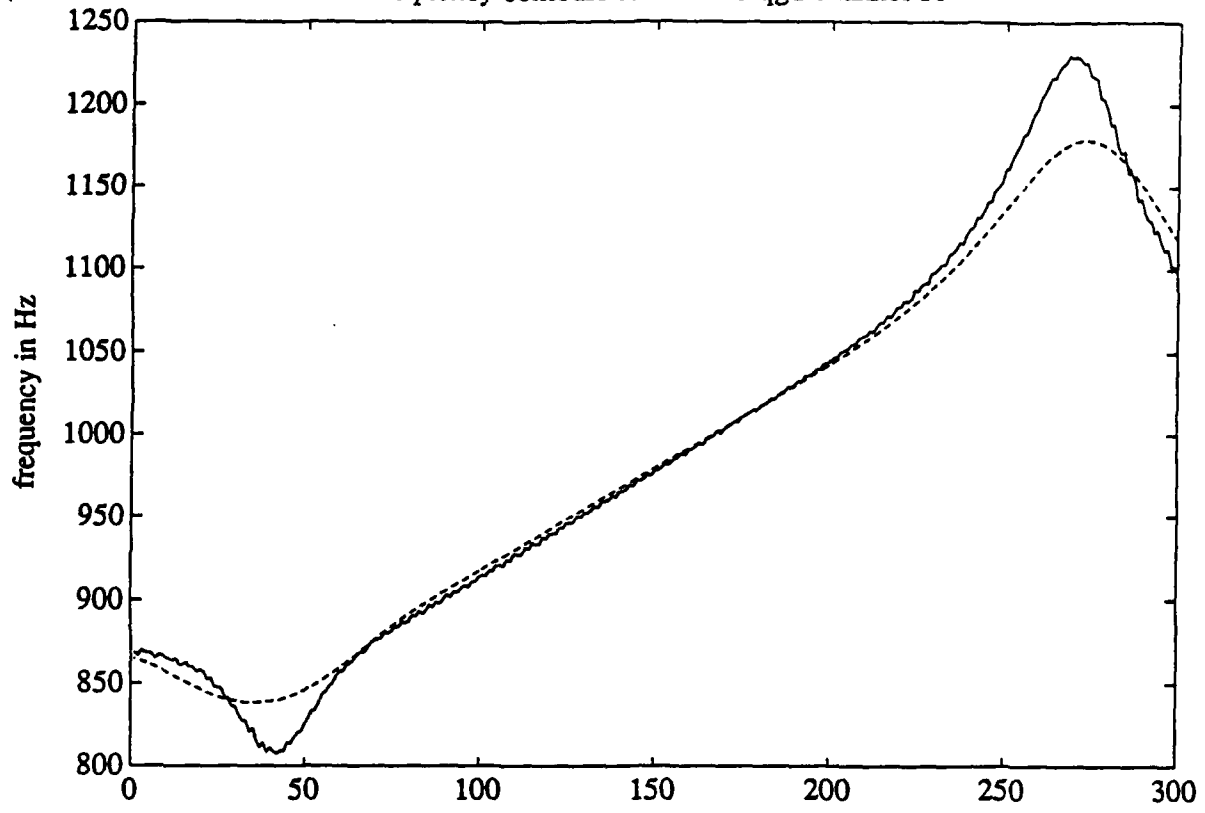
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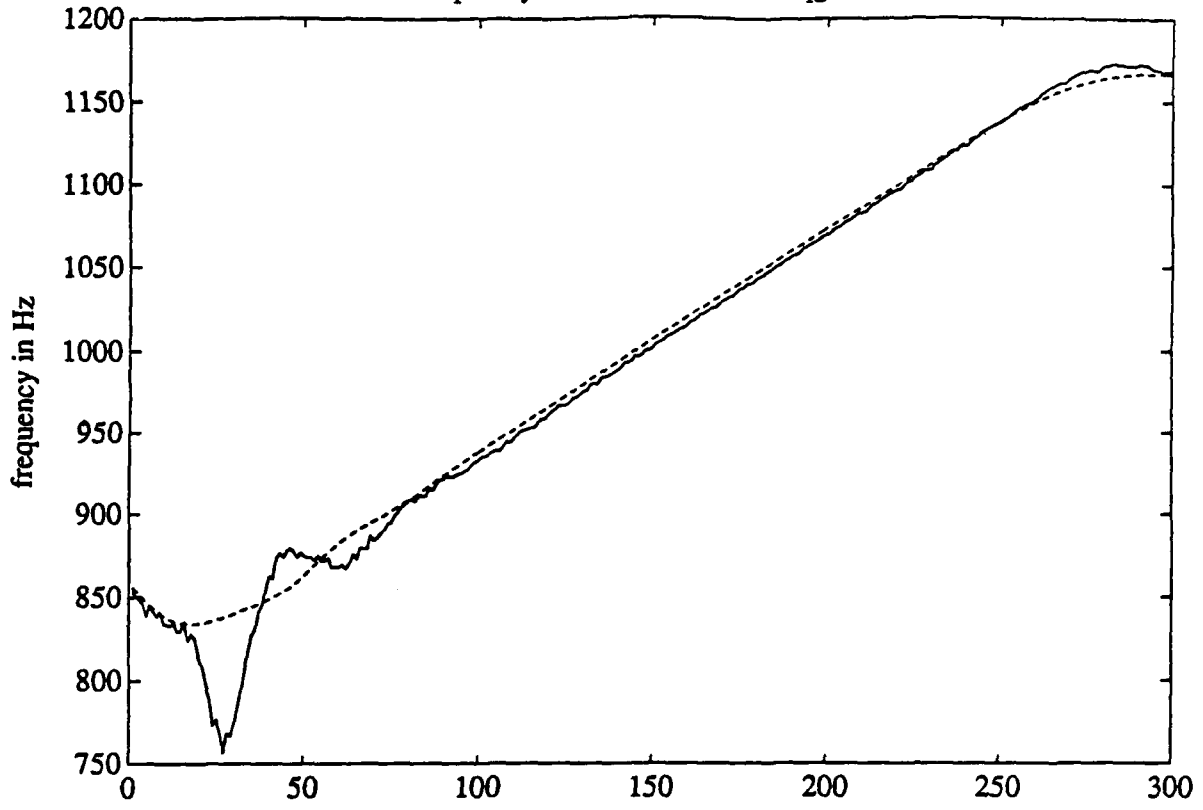
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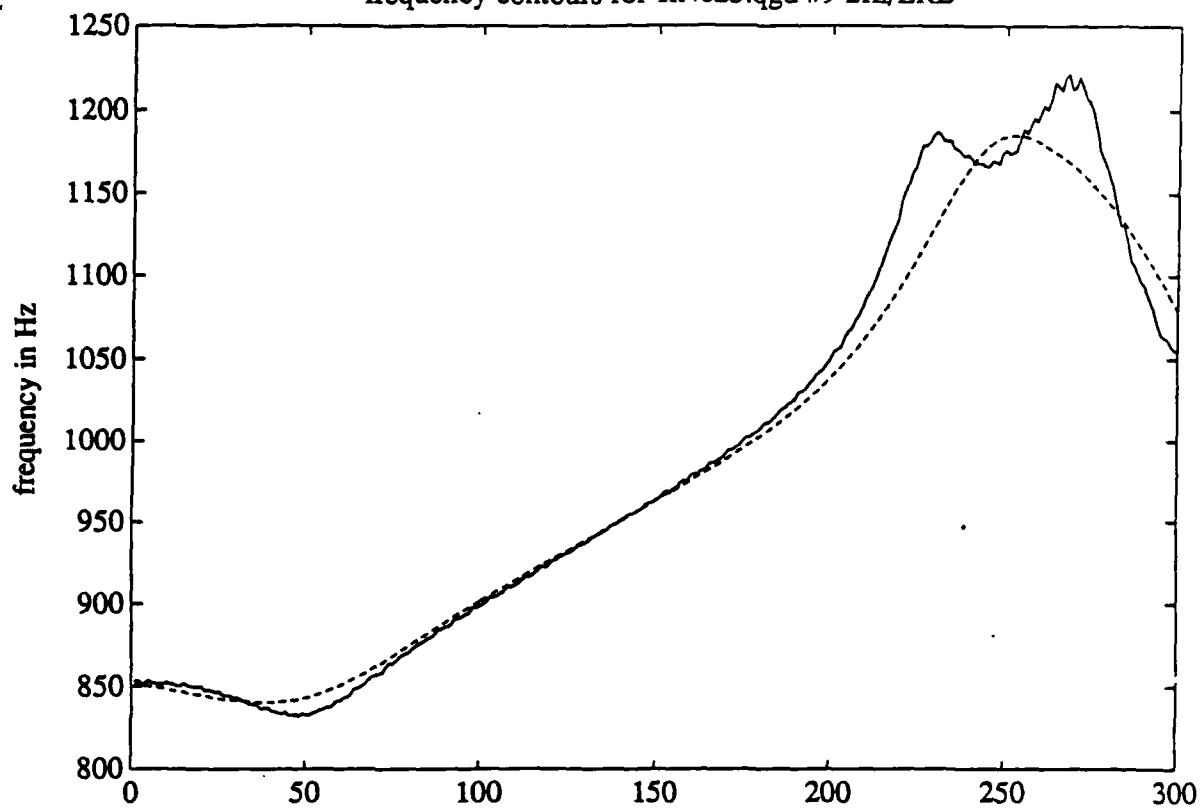
frequency contours for lk4c25.qgd channel 10



frequency contours for 1k4c25.qgd #11



frequency contours for 1k4c25.qgd #9 2fil/ERB



Vita

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Place of Birth: Madras, India

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EDUCATION:

Ph.D. in Electrical Engineering, University of Florida, Dec. 1983.

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B.Tech. in Electrical Engineering, Indian Institute of Technology, Madras, India, May 1979.

EXPERIENCE:

Assistant Professor, Department of Electrical Engineering, The Ohio State University, Columbus, Ohio, September 1986 - present.

Assistant Professor, Computer Science and Engineering Department, Auburn University, Auburn, Alabama, June 1984 - June 1986.

Visiting Assistant Professor, Department of Electrical Engineering, University of Florida, August 1983 - May 1984.

HONORS AND AWARDS:

Outstanding Engineering Faculty Award, Computer Science and Engineering Department, Auburn University, 1985-86.

SOCIETY MEMBERSHIPS:

Institute of Electrical and Electronics Engineers

- Computer Society

- Acoustics, Speech and Signal Processing Society

- Communications Society

Association for Computing Machinery

Acoustical Society of America

Eta Kappa Nu

PROFESSIONAL ACTIVITIES:

Reviewer, *IEEE Transactions on Acoustics, Speech and Signal Processing*.

Member, IEEE Computer Society Area Activities Board, 1985.

Vice-Chairman, Central Ohio Chapter of the Acoustical Society of America.

CONSULTING:

Summer 1986, National Semiconductor Corporation, Santa Clara, CA.

PUBLICATIONS:

Journal Articles

"A new competitive learning algorithm for vector quantization using neural networks," submitted to *Neural Networks*, November 1988, (with S. C. Ahalt, P. Chen, and D. E. Melton).

"A modified frequency-weighted Itakura spectral distortion measure," accepted for publication in *IEEE Transactions on Acoustic, Speech and Signal Processing*, (with J. Li).

"Two channel speech analysis," *IEEE Transactions on Acoustic, Speech and Signal Processing*, vol. ASSP-34, no. 4, pp. 730-743, August 1986, (with D. G. Childers).

"A critical review of electroglottography," *CRC Critical Reviews in Biomedical Engineering*, vol. 12, no. 2, pp. 131-161, 1985, (with D. G. Childers).

"Voiced Speech Model including Source-Tract Interaction," submitted to *Speech Communication*, September 1988, (with J. Li).

Conference Papers

"A new neural network learning algorithm for vector quantization," Microelectronics and Photonics in Communication Workshop, June 1989, (with S. C. Ahalt, P. Chen, and D. E. Melton).

"Performance of synthetic neural network classification of noisy radar signals," to appear in the *Proceedings of the IEEE Conference on Neural Information Processing Systems*, 1988, (with S.C. Ahalt, F.D. Garber, and I. Jouny).

"Formant Estimation and Vowel Recognition in Noise," Seventh Annual Conference, American Voice Input/Output Society, Oct. 1988, (with J. Li).

"Formant Estimation from Noisy Voiced Speech," 115th Meeting of the Acoustical Society of America, Journal of the Acoustical Society of America, Suppl. 1, 83, S1 (1988), (with J. Li and R. L. Moses).

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"Two channel (speech and EGG) analysis for formant tracking and glottal inverse filtering," in *Proceedings of the International Conference on Acoustics, Speech and Signal Processing*, San Diego, CA, March 1984.

"Glottal sensing for speech analysis and synthesis," in *Proceedings of the International Conference on Acoustics, Speech and Signal Processing*, Boston, MA, April 1983, (with J. J. Yea, J. M. Naik and D. G. Childers).

"Vocal fold vibratory patterns: Comparison of film and inverse filtering," in *Proceedings of the International Conference on Acoustics, Speech and Signal Processing*, Atlanta, GA, March 1981, (with D. G. Childers).

"Laryngeal function: Assessment and role in speech analysis and synthesis," in *Proceedings of the 10th International Congress of Phonetic Sciences*, Utrecht, The Netherlands, 1983, (with D. G. Childers, J. J. Yea and G. P. Moore).

"Spectral Analysis: AR, MA or ARMA," in *Proceedings of the 1st ASSP Workshop on Spectral Estimation*, August 1981, (with D. G. Childers and J. J. Yea).

Book Chapters

"Vocal source and tract models based on speech signal analysis," in *Mathematics and Computers in Biomedical Applications*, J. Eisenfeld and C. DeLisis, Eds., pp 335-349, Elsevier Science Publishers, 1985, (with D. G. Childers, E. L. Bocchieri and J. M. Naik).

"Electroglottography, speech and ultra-high speed cinematography," in *Vocal Fold Physiology*, I. R. Titze and R. C. Scherer, Eds., pp 202-220, The Denver Center for the Performing Arts, 1983, (with D. G. Childers, J. M. Naik, J. N. Larar and G. P. Moore).