2aPP14. A novel nonlinear system identification technique for modeling cochlear transduction using otoacoustic emissions data.
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A novel mathematical technique is presented for modeling the linear and nonlinear properties of the cochlear transduction process. This new nonlinear system identification procedure (Bendat, 1990) has several advantages over classical techniques used for analyzing nonlinear systems. With this technique, nonlinear systems may be modeled as a linear system in parallel with any number of finite-memory nonlinear systems. Higher-order nonlinear terms can be more easily computed with this technique than with others such as volterra kernel analysis. Furthermore, a broadband signal such as a transient signal can be used to characterize the cochlea over a wide frequency band. This makes the application of this procedure ideal for modeling cochlear transduction process with transient evoked otoacoustic emissions data. Otoacoustic emissions were recorded in response to finite impulse response (FIR) pulses from 11 subjects with normal hearing. System identification procedures were implemented using third- and fifth-order polynomial models. Based on the input and output records, system properties were estimated. Coherence measures tested the goodness-of-fit of the chosen model. The identified systems were used to predict the systems' response to two-tone stimuli. The fifth-order model predictions were in better accordance with previously reported data on acoustic distortion products.

2aPP15. Otoacoustic emissions and threshold shift correlations following asymptotic threshold shift producing noise exposures.
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Cubic distortion product emissions (3DPE) were measured in four groups of chinchillas ($N=4$ or 5/group) exposed for 5 days, 24 h/d, to 121-dB peak SPL, narrow-band (400 Hz) impacts having center frequencies (CF) of 1.0, 2.0, 4.0, or 8.0 kHz. The noise exposure produced an asymptotic threshold shift (ATS) at audiometric test frequencies showing a threshold shift (TS). Levels of ATS up to 80 dB were measured at and above the CF of the impact. Surprisingly, the 1.0- and 2.0-kHz CF impacts produced the largest ATS at 16 kHz. There was, in general, a congruence between the ATS and 3DPE changes following exposures to CFs $\geqslant 2.0$ kHz. However, with the 1.0-kHz CF impact exposure, the two metrics were not in general agreement. Relations among permanent thresholds shifts (PTS), 3DPEs and outer hair cell (OHC) losses were quite variable for low- and mid-frequency ($\leqslant 2.0$ kHz) biased lesions. For the 4.0- and 8.0-kHz CF impacts, 3DPE changes reflected OHC losses and PTS rather closely. DPEgrams using primary levels of 40 through 70 dB SPL in 10-dB increments were found to be most effective in the evaluation of OHC lesions than was any single DPEgram. [Research supported by NIOSH Grant No. R01 OH02317.]

2aPP16. An automated off-line procedure for detecting SOAEs from a power spectrum.
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Elimination of observer bias is an important advantage of automating the detection of spontaneous otoacoustic emissions (SOAEs), particularly when exploring differences in SOAE prevalence across subject populations. An objective procedure was developed for detecting peaks in the power spectrum, and estimating the likelihood that such peaks arose from SOAES. The procedure involved multiple passes. First, a smoothed estimate of the spectral noise floor was obtained by replacing spectral regions containing possible SOAEs with linear estimates extrapolated from adjacent frequency regions containing no peak deviations larger than a designated criterion, and then calculating a running average on that smoothed function. Potential SOAEs were then compared with this smoothed noise floor. Based on the variance of points in the original spectrum, the likelihood that any spectral peak did not occur by chance could be estimated. The efficiency and power of this procedure was explored for several combinations of smoothing procedures and criteria for eliminating suspected peaks in the first pass. The implications of varying the criterion for acceptance of a peak as an SOAE are discussed in terms of possible differences in SOAE prevalence in population samples. [Supported by NIDCD Grant DC 00153.]

2aPP17. Temporal aspects of pitch perception at high and low S/N ratios.
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The study investigated the integration of nonsimultaneous frequency components into a single pitch in quiet and in noise. Using a pitch matching paradigm, the shifts produced by a mistuned harmonic in the pitch of a target complex were measured either with a pre-target component (when the mistuned component stopped as the target started) or with a post-target component (when the mistuned component immediately followed the target tone) [J. Acoust. Soc. Am. **98**, 2946(A) (1995)]. The stimuli were presented in quiet or in the presence of a band of noise, low-pass filtered at a frequency cutoff of 3 kHz (approx. -23-dB S/N ratio around the frequency of the mistuned component). In quiet, post-target and simultaneous mistuned components produced larger pitch shifts than pre-target components. In noise, shifts in the pre-target condition were identical to those produced by simultaneous and post-target components. The duration of the interval which separated the nonsimultaneous components and the target tone was varied in a second experiment. The results suggest that the period over which nonsimultaneous components contribute to the pitch of the target is longer in noise than in quiet for both pre- and post-target conditions. [Work supported by HKRGC Grant No. HKU362/94M.]

2aPP18. Pitch identification in masking release paradigms.
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One of three pure-tone frequencies was presented in masking noise on a given trial, and the listener was asked to identify which pitch (low, middle, or high) was presented. The level of the signal varied adaptively. In a masking-level difference (MLD) paradigm, the noise was always interaurally in phase (No) and the signal was either interaurally in phase (So) or was interaurally inverted ($S\pi$). In a comodulation masking release (CMR) paradigm, a monaurally presented masking noise was either unmodulated or square-wave modulated at a rate of 30 Hz. In each paradigm, the parameter was the frequency separation between the signals. Detection thresholds were also measured for each frequency. When the frequency separation was relatively wide (e.g., 100 Hz or more), detection thresholds were similar to identification thresholds. As the frequency separation decreased, the identification thresholds increased steeply, particularly in the masking release conditions ($NoS\pi$ and modulated noise). Results will be discussed in terms of the different cues available in the baseline and masking release conditions, and in terms of why those cues may be less informative in identification than in detection conditions. [Work supported by NIDCD.]

2aPP19. Auditory short-term memory interference: Deutsch's demonstration revisited.
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The effect of intervening sounds on pure-tone pitch recognition was investigated. Intervening sounds were (1) none, (2) pure tones, and (3) naturally spoken numbers uttered by a female and a male speaker. Nine subjects who possessed no absolute pitch were tested for (1) pure-tone pitch recognition, (2) serial recall of the spoken numbers, and (3) both pitch recognition and number recall. GLM analysis was performed on the arcsine transformed error rates. It revealed that, when the intervening sounds were numbers, the pitch recognition error rates were significantly