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Speech dynamic range and its effect on cochlear implant performance

Submitted by

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Introduction

In this quarterly progress report, we update research activities occurring mostly in Hearing and Speech Research Laboratory at the University of California, Irvine (UCI). Since moving to UCI last summer, Dr. Zeng has completed his laboratory setup and personnel recruitment. The laboratory is now up and running. Here we present a manuscript that is ready for submission and report preliminary data on two additional studies conducted in Dr. Zeng's laboratory at UCI.

The first study was a collaborative effort with researchers at Johns Hopkins University, House Ear Institute, and Advanced Bionics Corporation, which measured speech dynamic range and its effect on cochlear implant performance. The main results are: (1) the speech dynamic range is about 50 dB, which is wider than the commonly-assumed 30 dB range; (2) a 50-dB input acoustic range is required to produce optimal speech recognition in cochlear implant users; and (3) further improvement in speech recognition may be achieved by implementing different acoustic-to-electric mapping functions for low- and high-frequency channels.

The second study focused on the effectiveness of cochlear implantation in auditory neuropathy. People with auditory neuropathy typically preserve the cochlear amplification function (presence of otoacoustic emission) but have desynchronuous neural activities (absence of evoked auditory brainstem responses). They often complain about hearing but not being able to understand sounds, particularly in noise. Because hearing aids are usually ineffective, cochlear implantation has been attempted to alleviate the hearing problem inthis group of people. Here we present psychophysical and electrophysiological data in two neuropathy patients who have received cochlear implants. While the results showed that electric stimulation significantly improves neural synchrony, the neuropathy patients with cochlear implants still cannot reach the level of temporal processing by a typical cochlear implant user. We propose that a slower rate but more channels of stimulation may be more beneficial to the neuropathy population.

The third study was aimed to identify the critical information necessary for accurate perception of music and tonal languages. Our preliminary data showed that cochlear implant users can typically recognize tempo and rhythmic patterns but cannot identify commonly known melodies. These results suggest that cochlear implant listeners have relatively normal temporal processing but impaired processing of fine-frequency structure. To achieve a high level of musical appreciation, this fine-frequency structure has to be encoded in future cochlear implants.

Research activity update

The Hearing and Speech Research Laboratory at UCI occupies a 1000-square-foot space in the College of Medicine. The laboratory has two double-walled, sound-attenuated booths equipped with modern digital sound generation and delivery systems (Tucker-Davis System II and III). Clinical and research interfaces are also available for all cochlear implant devices (Clarion, Ineraid, Med-El, and Nucleus). The laboratory is in close proximity to the Evoked Potentials Laboratory (Dr. Starr), the Auditory Neurophysiology laboratory (Dr. Kitzes), and the Brain Imaging Center. The following is a summary of research activities that have occurred since Dr. Zeng's move to UCI.

Laboratory personnel:

- Research Associate: Rachel Cruz, M.A (Northwestern University), CCC-A; Ms. Cruz has a background in music and audio engineering.
- Post-doc: Ginger Stickney, Ph.D. in Psychology (University of Texas – Dallas), CCC-A; Dissertation title: "Analyses of speech processing strategies for cochlear implants and the effects of electrode interaction."
- Post-doc: Kaibao Nie, Ph.D. in Biomedical Engineering (Tsinghua University); Dissertation title: "Speech signal processing for cochlear implants."
- Doctoral Student: Sheng Liu, M.S., Department of Biomedical Engineering, UCI. Mr. Liu has a background in biomedical signal processing.
- Doctoral student: Yingyee Kong, M.S., Department of Cognitive Sciences, UCI. Ms. Kong has a background in linguistics and psychophysics.
- Doctoral student: Zhongqiang Ding, M.S., Department of Information and Computer Sciences, UCI. Mr. Ding has a background in programming and speech coding.

Research activities:

- Ruth Litovsky, Ph.D., from Boston University, was brought in as a consultant to prepare a study on binaural hearing for patients who have received bilateral cochlear implants.
- Representatives from Advanced Bionics, Med-El, and Cochlear Corporation visited UCI to continue their technical support for developing improved speech processing strategies for auditory prostheses.
- Zeng visited Janet Shanks and Lisa Gibbs at Long Beach VA Hospital to establish collaboration and access to VA implant subjects.
- Cruz, and Stickney and Zeng visited Oralingua School for the Hearing Impaired at Whittier, California.
- Sheng Li and Zeng attended the Clarion C-II research interface workshop
- Zeng gave talks about cochlear implants at UCSF, UC-Berkeley, and the University of Michigan, Ann Arbor.

Speech dynamic range and its effect on cochlear implant performance

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Abstract
This report studies how to optimally convert speech sounds into audible electric currents in cochlear-implant listeners. The speech dynamic range was measured using twenty consonants and twelve vowels spoken by five female and five male talkers. Under conditions where the maximal rms level was normalized for all phoneme tokens, both broad-band and narrow-band acoustic analyses showed about a 50-dB distribution of envelope levels. This 50-dB speech dynamic range has to be mapped into a 10-20 dB electric range, which is typical in cochlear implant users. Using a logarithmic mapping function, speech recognition was evaluated in ten Clarion implant users as a function of the input acoustic dynamic range. The recognition data showed that a 50-dB input dynamic range is required to produce optimal speech recognition in these implant users. Taken together, the present acoustic and perceptual data indicate that the speech dynamic range is much greater than the commonly-assumed 30-dB range. A new amplitude mapping strategy is proposed based on the acoustic analysis of the envelope distribution difference between consonants and vowels. This new strategy uses a logarithmic map for low-frequency channels and a more compressive map for high-frequency channels, and may improve overall speech recognition for the present cochlear implant users.

INTRODUCTION

A major goal in designing speech processors for cochlear implants is to optimally convert speech signals into electric currents that fit in the implant user's perceptual range. In order to make the softest speech sounds audible and the loudest still comfortable, we need to know the dynamic range for speech sounds, the dynamic range for electric stimulation, and the appropriate conversion from speech sounds to electric currents. In clinical practice, selection of acoustic and electric dynamic ranges and conversion from acoustic amplitude to electric amplitude are part of the "mapping" process, which can play an important role in determining the outcome of cochlear implant performance and satisfaction. Psychophysical studies have measured the dynamic range over a large electric parameter space and determined the appropriate conversion from acoustic amplitude to electric amplitude (e.g., Zeng and Shannon, 1992; 1994; 1995; Zeng, Galvin, and Zhang, 1998; Zeng and Galvin, 1999). However, much less is known about how much speech information should be included in the input dynamic range for cochlear implants. Here we present new empirical data on the speech dynamic range and demonstrate its significance in cochlear implant performance.

Ideally, the input dynamic range would be set to 120 dB, the typical dynamic range within which a normal-hearing person processes acoustic intensity information. The acoustic amplitude with the 120-dB dynamic range would then be converted into a current value that evokes sensation between minimal to maximal loudness. However, the acoustic dynamic range has to be greatly compressed to accommodate the substantially narrow dynamic range for the cochlear implant listeners (about 10-20 dB, see Skinner et al., 1997; Zeng and Galvin, 1999, and Table 2 in this study). Because the implant listeners also have a limited intensity resolution of about 20 discriminable steps (Nelson et al., 1996; Zeng et al., 1998), they would not be able to discern meaningful variations in sound intensity. Practically, speech is likely the most important sound and usually has a much smaller dynamic range than the 120-dB range, therefore, the input dynamic range is set between 30 and 60 dB in most implant devices. The hope is that the relative intensity changes from soft consonants to loud vowels will be preserved perceptually for a cochlear implant listener to understand speech.

Currently, there are more than 40,000 cochlear implant users worldwide. Nearly three quarters use the Nucleus device by Cochlear Corporation. In Nucleus devices, a 30-dB range is used as the input dynamic range (User Manual, The Nucleus 22 Channel Cochlear Implant System, p. 4-SP). In Med-El devices, a 60-dB input dynamic range is used (Stobich et al., 1999). In Clarion devices, the input dynamic range can between 20 and 80 dB for users of the Simultaneous-Analog-Stimulation (SAS) strategy (Clarion Device Fitting Manual, C9055003-002 Rev. C, p.220). At present, the clinical fitting of the acoustic dynamic range relies mostly on experience and lacks experimental validation.

The 30-dB speech dynamic range is widely assumed, based on the classic acoustic analysis by Fletcher (1953) and other earlier statistical measurements on conversational speech (Dunn and White, 1940). This 30-dB dynamic range has formed the basis for many applications including the Articulation Index (ANSI, 1969; 1997). However, modern analysis using digital signal processing has shown a much greater speech dynamic range than this classic 30-dB range. Boothroyd et al. (1994) performed one-third octave analyses of 7 phonemes produced by 5 female and 5 male talkers. They found that the overall dynamic range in these data was 53 dB, and that the dynamic range was 37 dB even after adjustment in overall levels and high-frequency pre-emphasis. Stobich et al. (1999) calculated the distribution of envelope level for 180 German sentences spoken by a male talker and found a dynamic range of 70 dB for these speech materials. Eddington et al. (1999) also calculated the distribution of envelope levels over 6 frequency channels for the TIMIT sentences presented at a conversational level. They found that the distribution of speech envelope levels was in the range of 40-60 dB.

Perceptual studies also support the data from modern acoustic analysis that the speech dynamic range is greater than 30 dB. Studebaker et al. (1999) measured NU6 word recognition at speech levels from 64 to 99 dB SPL and speech-to-noise ratios from -29 to -4 dB. They found a slight increase in speech recognition scores (5 rau units) when the speech level was increased from 64 to 79 dB SPL. This suggests that, contrary to the 30-dB speech dynamic range assumption, audibility was still increasing under these conditions. Moreover, if the 30 dB dynamic range were assumed, then the lowest amplitudes for the speech sound would be 15 to 18 dB lower than the speech rms level, as assumed in ANSI (1969 and 1997). In other words, word recognition for speech-to-noise ratios ranging from 16 to 28 dB should be similar to that produced by the quiet condition. However, Studebaker et al. found significantly poorer speech recognition for the noise conditions than the quiet condition. The speech score decreased by 5 to 15 rau units when the speech-to-noise ratio was varied from 16 to 28 dB with speech presented at a fixed 65-dB SPL overall level. The speech score was further decreased by 5 to 25 rau units for the same speech-to-noise ratios with speech presented at a fixed 99 dB SPL overall level. These results led Studebaker and his colleagues to conclude that the effective dynamic range of speech must be at least 40 to 43 dB.

Here we measured the distribution of envelope levels for two widely-used speech test materials: 12 vowels in /hVd/ format (Hillenbrand, Getty, Clark, & Wheeler, 1995) and 20 consonants in /aCa/ format (Turner, Souza, & Forget, 1995; Shannon et al., 1999). Our data showed that these speech materials have a 50-dB envelope level distribution based on either a broad-band analysis or a narrow-band analysis from 8 frequency channels. We also measured speech recognition as a function of input dynamic range in cochlear implant listeners. Our data showed that an input dynamic range of 50-60 dB is required to produce optimal performance for cochlear implant users.

I. Methods

A. Subjects

Five young (21-36 years old) normal-hearing listeners served as a control in the experiment. Ten Clarion™ (Advanced Bionics Corporation) cochlear implant users also participated in the experiment. The implant subjects' ages ranged from 21 to 56 years (average 42 years). Each subject had at least one year of experience with the cochlear implant during the time of testing. There were 7 CIS users and 3 SAS users. Except for one subject (MY) who was deafened prelingually, all other subjects had post-lingual deafness. All subjects were familiar with speech tests from previous clinical evaluations. Additional subject information is listed in Table 1. Local IRB-approved informed consent was obtained. All subjects were paid for their participation.

Insert Table 1 about here

B. Clarion Speech Processors

Each cochlear implant listener used his or her preferred clinical setting (or map) for the experiment. User maps were uploaded from the subject's speech processor, stored in SCLIN for Windows software (Clarion Device Fitting Manual), and downloaded to a laboratory S-Series speech processor to minimize equipment-related variables. Speech recognition was conducted as a function of the input dynamic range (IDR). There were 6 possible settings with the CIS processing strategy (from 10 to 60 dB in 10 dB steps) and 7 possible settings with the SAS processing strategy (from 20 to 80 dB in 10 dB steps). No changes other than the IDR were made within an individual's map. Volume and sensitivity settings were kept constant within and between test sessions.

Figure 1 illustrates the detailed relationship between input dynamic range and electric dynamic range in Clarion cochlear implants. The x-axis (i.e., the input dynamic range) determines the range of acoustic input mapped into the electric output range between threshold (T level) and the most comfortable loudness (M level). The speech processor first selects an acoustic level (0 dB on the x-axis) and maps it into an electric level (M level) that evokes the most comfortable loudness. The speech processor then maps either the 10 dB range below the 0-dB acoustic level into the audible electric dynamic range (the rightmost sloping line), or any other acoustic range into the same audible electric dynamic range. Presumably, any acoustic input level that is outside the input dynamic range will be mapped into either a subthreshold electric level (< T level) or a constant saturating level (> M level). Note the interchangeable relationship between the input dynamic range and the T level. For example, a 40-dB input dynamic range setting effectively becomes a 20-dB setting when the electric threshold is increased from T level to T' level (see 2 open circles in Fig. 1). Because the x-axis is logarithmic while the y-axis is linear, a straight line on these axes indicates a logarithmic compression from acoustic amplitude to electric amplitude. This logarithmic transformation between acoustic and electric amplitude has been verified psychophysically to restore normal loudness growth in electric stimulation (Eddington et al., 1978; Zeng and Shannon, 1992; Dorman et al., 1993).

Insert Fig. 1 about here

C. Stimuli

Five female and five male adult talkers produced 12 vowels in /hVd/ format (Hillenbrand, Getty, Clark, & Wheeler, 1995) and 20 consonants in /aCa/ format (Turner, Souza, & Forget, 1995; Shannon et al., 1999). The Hillenbrand vowels were 16-bit .WAV files sampled at 16 kHz, and the Turner/Shannon consonants were 16-bit .WAV files sampled at 44.1kHz. All speech tokens (including Hillenbrand vowels and

Turner/Shannon consonants) were subject to a normalization procedure based on the maximal rms level from a 50-ms running window. This maximal level most likely measured the level of the steady-state portion of the vowel.

These vowel and consonant stimuli were output via a PC soundcard (Turtle Beach MultiSound Fiji board) connected to one channel of a mixer (Tucker-Davis Technologies, TDT SM1). A speech-spectrum-shaped noise was generated by passing white noise (TDT WG1) through a specially-designed low-pass filter with a cut-off frequency at 608 Hz and a -12 dB/octave slope (Byrne et al., 1994). The noise was delivered to another channel of the mixer where it was summed with the phonemic stimuli.

The summed speech and noise stimuli were amplified (Crown D-75) and presented to the listener via a Tannoy Reveal speaker mounted on a double-walled sound-treated booth (IAC). Each subject was positioned in the center of a double-walled sound-treated room (IAC) facing the speaker (about 1 meter away, at 0° azimuth and at ear level). A calibration vowel /a/ was generated to have the same rms level as the average vowel level in both tests and to produce a conversational level of 65 dBA. The noise was attenuated (TDT PA4) to achieve a +5 dB speech-to-noise ratio (i.e., the noise had a level of 60 dBA).

D. Procedures

Distribution of speech envelope levels was calculated for both broad-band (250-6800 Hz) and narrow-band analysis. In the broad-band analysis, the envelope of the acoustic signal was extracted by full-wave rectification and low-pass filtering (an Elliptical IIR filter with 160-Hz cutoff frequency and –6 dB per octave slope). A histogram was calculated to produce the number of occurrences for envelope amplitude (re: peak amplitude). Because of the noise floor on the bottom of the distribution, we conservatively defined the speech dynamic range as the difference in the envelope levels producing between 5% and 99% accumulative occurrences. In the narrow-band analysis, the broad-band signal was divided into 8 narrow bands (Fourth order Elliptical IIR filters with cutoff frequencies at 250, 500, 875, 1150, 1450, 2000, 2600, 3800, and 6800 Hz). These filters corresponded to the filters used in the Clarion cochlear implants. The band-specific envelope was extracted and its amplitude histogram was constructed in the same way as the broad-band analysis.

Vowel and consonant recognition were conducted separately in a closed-set format using an interface developed at the House Ear Institute (Robert, 1999). The test order of different input dynamic ranges was pseudo-randomized for all listeners. Speech recognition was conducted first in quiet and then in noise. All listeners were given 15 minutes to acclimate to each experimental processor and were allowed to preview all stimuli before formal test sessions. Each test session consisted of 5 presentations for each phoneme by each of the ten talkers. The order of each phoneme's occurrence in each test session was randomized. The listener's response to the speech stimulus was stored as a confusion matrix. No trial-by-trial feedback was given regarding the correctness of the response.

II. Results

A. Speech dynamic range

Figure 2 shows distribution of envelope levels for these /aCa/ and /hVd/ tokens in the broad-band condition (top panel) and for the /aCa/ tokens (middle panel) and the /hVd/ tokens (bottom panel) in the 8-channel condition. First note the dominating envelope level distribution at high levels for the broad-band analysis. A small "bump" in the distribution at low levels (more obvious in the vowel envelope) most likely reflects the contribution from the soft consonants. This is clearly illustrated in the narrow-band analysis, which shows a strong distribution at low levels for the high-frequency channels (dotted lines in middle and bottom panels).

For the broad-band condition (top panel), the acoustic dynamic range is 47 dB (from -51 to -4 dB) for consonants and 46 dB (from -50 to -4 dB) for vowels. For the 8-channel condition, the consonant dynamic range is 41, 52, 51, 50, 47, 46, 47, and 45 dB for channel 1, 2, 3, 4, 5, 6, 7, and 8, respectively. On the other hand, the vowel dynamic range is 51, 51, 53, 49, 47, 47, 42, and 36 dB for channel 1, 2, 3, 4, 5, 6, 7, and 8, respectively. Given these acoustic dynamic ranges, we shall see whether an input dynamic range setting of 50 dB would produce optimal speech recognition in cochlear implant users.

Insert Fig. 2 about here

B. Electric dynamic range

Figure 3 shows the most comfortable loudness (M levels, top panel) and the threshold (T levels, bottom panel) as a function of electrode position in 5 CIS and 3 SAS users. These M and T levels are presented in microamps. Note the apparent;y greater variability in both M and T levels for the SAS users compared to the CIS users. Also note the greater variability across electrodes for the SAS users than the CIS users. The high M levels for subject AL may actually be much lower as they approach the saturation portion of the current source in the Clarion S-series devices (Clarion Device Fitting Manual p. 20).

Insert Fig. 3 about here

Table 2 shows the calculated electric dynamic ranges, defined as the dB difference between M and T levels. Table 2 confirms the visual impression in Fig. 3 that the SAS users have both greater across-subject and within-subject variability in dynamic range than the CIS users. The electric dynamic range averaged across electrodes was 15.0, 13.9, 13.0, 11.6, and 12.8 dB for the CIS users, LH, JM, MY, SC, and WC, respectively. On the other hand, the averaged electric dynamic range was 44.6, 6.6, and 9.4 dB for the SAS users, AL, AM, and EC, respectively. Note that the unusually large dynamic range for AL is a theoretical value, the actual value may be much lower and could be calculated, if we had access to the subject's internal device. Similarly, the variability in dynamic range within the subject is much smaller (standard deviation ranges from 0.2 to 1.6 dB) for the CIS users than the SAS user (standard deviation ranges from 0.8 to 10.0 dB).

Insert Table 2 about here

C. Phoneme recognition in quiet

Figure 4 shows both the group average (line) and individual data (symbol). The top panel shows consonant recognition (y-axis) as a function of input dynamic range (x-axis), while the bottom panel shows vowel recognition (y-axis) as a function of input dynamic range (x-axis). For the 5 normal-hearing listeners, the average score for consonant recognition was 97% and the score for vowel recognition was 93%. For the implant listeners, the best average score was about 40 percentage points lower than the normal-hearing control; even the best individual score was still about 15 percentage points lower than the control.

More interestingly, the average group data demonstrated a non-monotonic function with the best performance at medium input dynamic ranges of 40-60 dB and a decreased performance at lower and higher input dynamic ranges. The individual data had a similar trend, but their range of performance varied greatly. The individual performance range was between 30 and 45 percentage points for all except the -70 and -80 input dynamic range conditions, at which only 1-2 subjects participated in the experiment.

A one-way ANOVA confirmed that the input dynamic range is a significant factor affecting speech recognition in Clarion cochlear implant users [consonants: $F(7,36)=6.19$, $p<0.01$; vowels: $F(7,33)=2.79$, $p<0.05$]. A paired t-test indicated no significant difference in consonant recognition between the -50 and the -60 dB input dynamic range conditions ($p>0.05$), but significantly poorer performance for the remaining narrower input dynamic range conditions ($p<0.01$). Similarly, there was no significant difference between -40, -50, and -60 dB conditions ($p>0.05$) in vowel recognition, which was significantly better than the -10, -20, and -30 dB input dynamic range conditions. No statistical test was conducted between the medium dynamic range and the -70 and -80 dB conditions because of the small number of subjects. The present data suggest that the input dynamic range should be set to 50 dB or greater in order to achieve optimal performance in speech recognition.

Insert Fig. 4 about here

D. Phoneme recognition in noise

Figure 5 similarly shows consonant and vowel recognition as a function of input dynamic range for the 5-dB speech-to-noise ratio condition. For comparison, the averaged data for the quiet condition are shown as the dashed line. Because not all subjects were tested in every condition, the averaged data for the quiet condition in Fig. 5 are shown for only those conditions where the corresponding noise data were available. Because of the scarcity of the data, we only performed a paired t-test, which revealed that noise significantly lowered both consonant recognition scores ($p<0.001$) and vowel recognition scores ($p<0.01$). A closer examination on the pattern of data in Fig. 5 also revealed a couple of interesting trends. First, the noise seemed to “flatten” both consonant and vowel recognition functions. This trend was particularly apparent with vowel recognition. Second, noise appeared to affect consonant recognition more with wide input dynamic range settings (a decrease of 20 percentage points for input dynamic ranges between 50 and 80 dB) than with narrow input dynamic range settings (merely a decrease of 4 percentage points for the 30 dB input dynamic range).

Insert Fig. 5 about here

III. DISCUSSION

The present acoustic analysis and perceptual results can shed light on how to optimally map speech dynamic range into electric dynamic range. The acoustic analysis results showed that multi-talker phonemes have approximately a 50-dB distribution of envelope levels, which is much wider than the commonly-assumed 30-dB speech dynamic range. In the broad-band (250-6800 Hz) analysis, the distribution of consonant and vowel envelope levels, particularly the vowel levels, showed a bi-modal pattern (top panel in Fig. 2). This bi-modal distribution disappeared in the narrow-band analysis (middle and bottom panels in Fig. 2), approximating a normal distribution with different means for different frequency bands. The high-frequency channels have a shifted distribution towards lower envelope levels than the low-frequency channels. Presumably, the high-frequency channels carry most consonant information such as fricatives and stops, while the low-frequency channels carry mostly vowel information. This difference in envelope level distribution can significantly affect how consonants and vowels should be mapped into an audible electric range.

Acoustic-to-electric amplitude mapping has been studied extensively in users of auditory brainstem implants (Shannon, Zeng, and Wygonski, 1992), the Med-El/CIS-Link Ineraid devices (Boex et al., 1995; Wilson et al., 1999; Loizou, Poroy, and Dorman, 2000), and the Nucleus devices using either 4-channel CIS-type processing (Fu and Shannon, 1998) or the SPEAK strategy (Zeng and Galvin, 1999). A general trend noted in these studies was that a more compressive map would produce better consonant recognition than a less compressive map, while the degree of compression has little, if anything at all the opposite, effect on vowel recognition (Boex et al., 1995; Zeng and Galvin, 1999). The present acoustic analysis can account for this observation.

Figure 6 shows a case where the acoustic envelope amplitude of both consonants (dotted line) and vowels (solid line) is mapped into the electric level using the same logarithmic function (assuming input dynamic range is in dB and electric level is in microamps). The two horizontal dashed lines represent the electric threshold (T level) and the most comfortable loudness (M level), respectively. Because the consonant envelope distribution was about 20 dB lower than the vowel envelope distribution (Fig. 2 middle and bottom panels), the consonants are likely to be mapped into a less optimal electric range. First, some low envelope levels may be mapped into electric levels below threshold (the lower horizontal dotted line). Second, a portion of the electric dynamic range may be wasted (the portion indicated by the line with an arrow on both ends) because no envelope levels are present. Third, even envelope levels that are mapped into audible electric range, are likely mapped into the lower portion of the electric dynamic range where intensity discrimination and modulation detection are both poor (Nelson et al., 1996; Zeng et al., 1998; Fu, 2000).

Insert Fig. 6 about here

On the other hand, if a more compressive map is used for consonants, then all three undesirable effects can be alleviated. Figure 7 shows the same map as in Figure 6 for vowels but a more compressive map for consonants (the curved line on right-top panel). The compression will raise previously inaudible low envelope levels above threshold, reduce the unused portion of the electric dynamic range, and map more of the envelope into the upper electric dynamic range where intensity discrimination and modulation are optimal. The negative trade-offs for the more compressive mapping are the slightly distorted envelope level distribution (see the mapped consonant envelope distribution in electric domain, left-top panel) and the possibility that some low-level noise may also become audible. Overall, a more compressive map is likely to produce better consonant recognition than a less compressive map, as seen in the literature.

Insert Fig. 7 about here

Theoretically, under laboratory conditions where the envelope level distribution for test materials is known, one can optimally set each channel's mapping function based on the mean and standard deviation of the envelope level distribution of that channel. Under realistic listening situations where speech materials cannot be controlled and real-time processing is required, more compressive mapping for high-frequency channels relative to low-frequency channels will help map the consonant envelope levels into the full electric dynamic range. In other words, cochlear implant users may achieve better overall speech recognition with a logarithmic map for low-frequency channels and a more compressive map for high-frequency channels. Such implementation is not feasible with the present clinical fitting systems. A future study using a research interface is required to implement the different mapping functions for different frequency channels and to evaluate its predicted improvement in speech recognition.

IV. CONCLUSIONS

The present study measured the speech dynamic range using twenty consonants and twelve vowels spoken by five female and five male talkers. The present study also measured speech recognition in Clarion implant users as a function of the input acoustic dynamic range. The acoustic and perceptual data support the following conclusions:

1. The speech dynamic range is about 50 dB, much wider than the commonly-assumed 30 dB dynamic range.
2. An input dynamic range of 50-60 dB is required to support optimal speech recognition in cochlear implants.
3. Current cochlear implant users may benefit from a new amplitude mapping strategy where a logarithmic map is used for low-frequency

channels and a more compressive map is used for high-frequency channels.

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Table 1. Biographical and audiological information for cochlear implant participants in this study.

Subject	**Age**	**Surgery Date**	**Device**	**Strategy**	**Ear**	**Etiology**
DF	66	11/16/89	S-Series	CIS	Left	Otosclerosis
JM	39	7/16/97	S-Series	CIS	Left	Unknown
LH	21	8/5/98	S-Series	CIS	Right	Unknown
SC	56	4/25/96	1.2	CIS	Left	Meningitis
MY	56	11/20/96	1.2	CIS	Left	Maternal Rubella
NJ	55	1/17/97	S-Series	CIS	Right	Congenital
WC	35	12/5/96	1.2	CIS	Right	Ototoxicity
AL	46	1/29/98	S-Series	SAS	Left	Unknown
AM	61	5/16/97	S-Series	SAS	Right	Menieres
EC	76	7/9/98	S-Series	SAS	Right	Unknown

Table 2. Electric dynamic range (dB) for 5 CIS users (LH, JM, MY, SC, and WC) and 3 SAS users (AL, AM, and EC).

MY	WC	JM	SC	LH	AL	AM	EC
11.9	13.6	12.8	10.1	13.1		6.1	23.3
14.8	13.9	13.4	11.9	12.8	38.7	7.5	24.8
14.2	11.3	13.1	12.2	12.8	41.3	6.9	3.7
14.5	13.6	13.1	12.5	12.8	41.6	7.5	2.9
17.1	13.9	12.5	12.8	13.1	45.1	6.4	3.2
16.8	14.5	13.1	11.9	12.8	50.6	6.6	2.9
14.8	15.1	13.6	11.3	12.8	50.3	5.2	4.9
15.7	15.4	12.2	10.2	12.5			
Average:							
15.0	**13.9**	**13.0**	**11.6**	**12.8**	**44.6**	**6.6**	**9.4**
Std Dev:							
1.6	**1.2**	**0.5**	**1.0**	**0.2**	**5.0**	**0.8**	**10.0**

FIGURE LEGENDS:

Figure 1. Conversion from input dynamic range (dB, x-axis) to electric dynamic range (μA, y-axis) in Clarion devices. M level is the most comfortable loudness level. T level represents electric threshold. Raising T level has the same effect as narrowing the input dynamic range (e.g., raising threshold from T to T' effectively reduces the input dynamic range from –40 to –20 dB, see the 2 open circles).

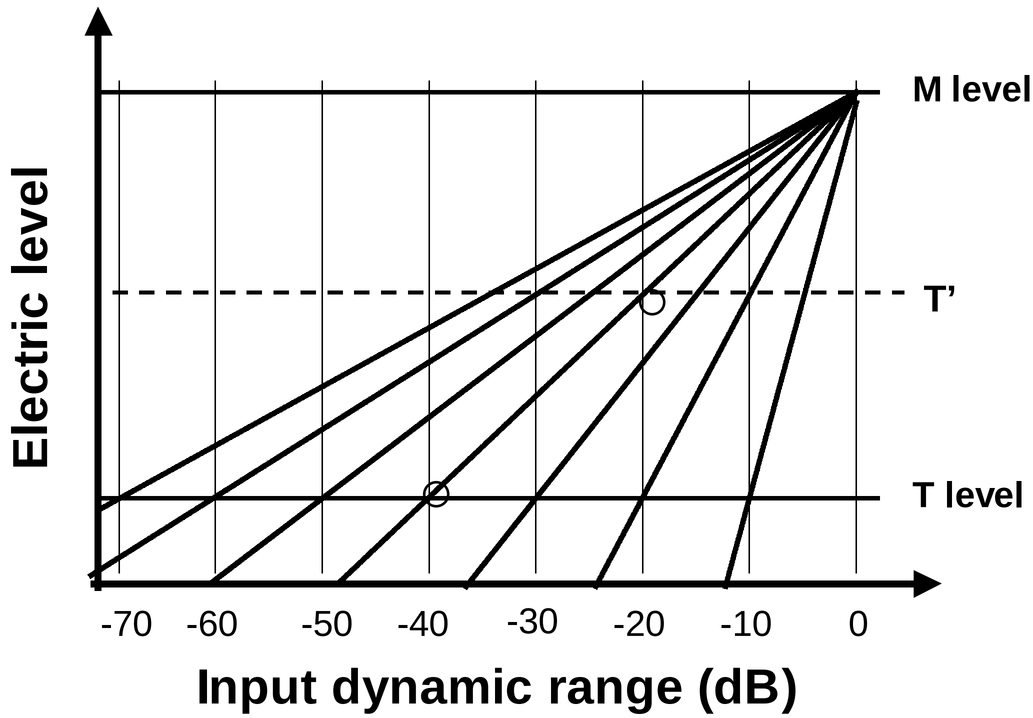


Figure 2. Speech dynamic ranges (or envelope level distributions) for the broad-band condition (top panel) and the 8-narrow-band conditions (for consonants see the middle panel and for vowels see the bottom panel).

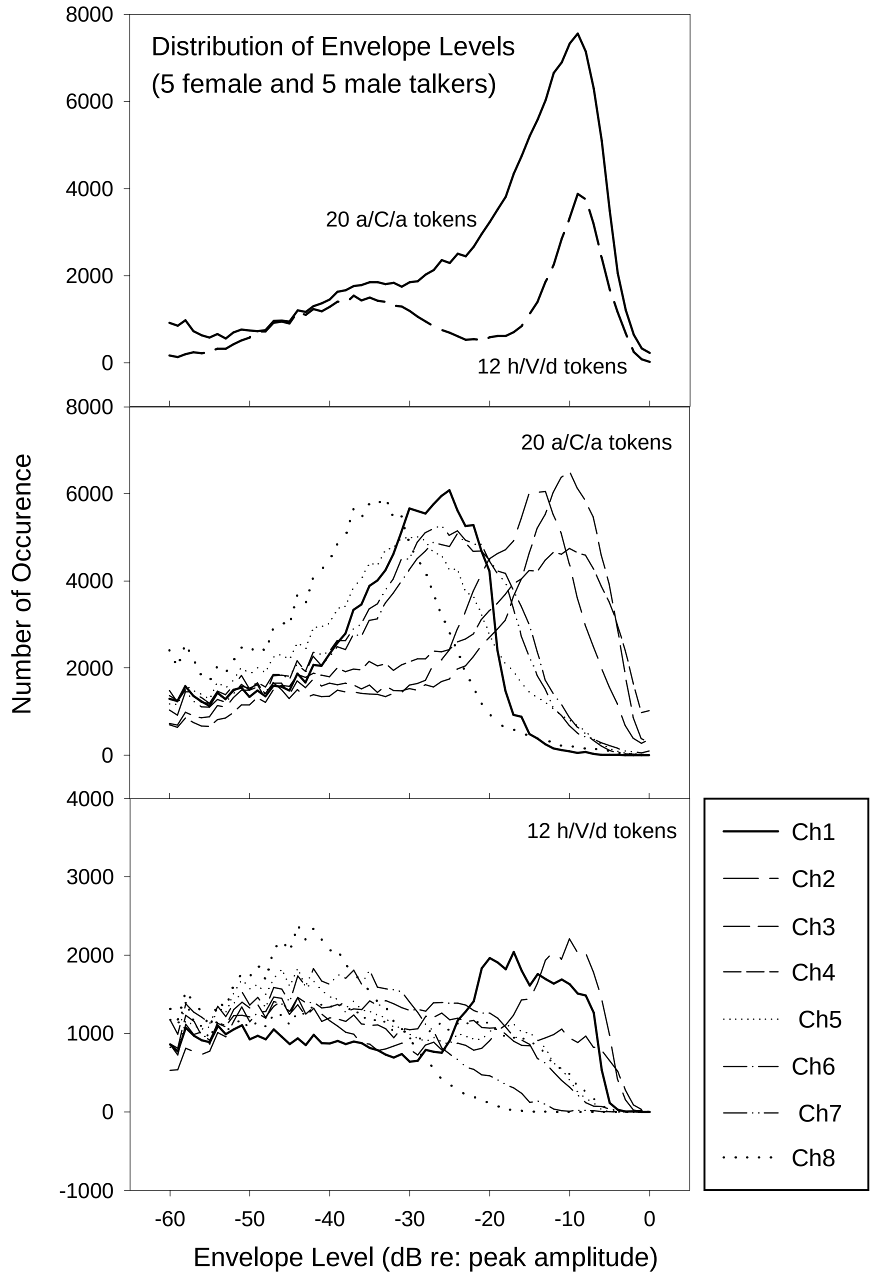


Figure 3. Top panel displays the most comfortable loudness (M-level) as a function of electrodes (x-axis). Bottom panel displays thresholds (T-level) as a function of electrodes (x-axis). Note the greater variability among the SAS users (filled symbols connected by solid lines) than the CIS users (open symbols connected by dashed lines).

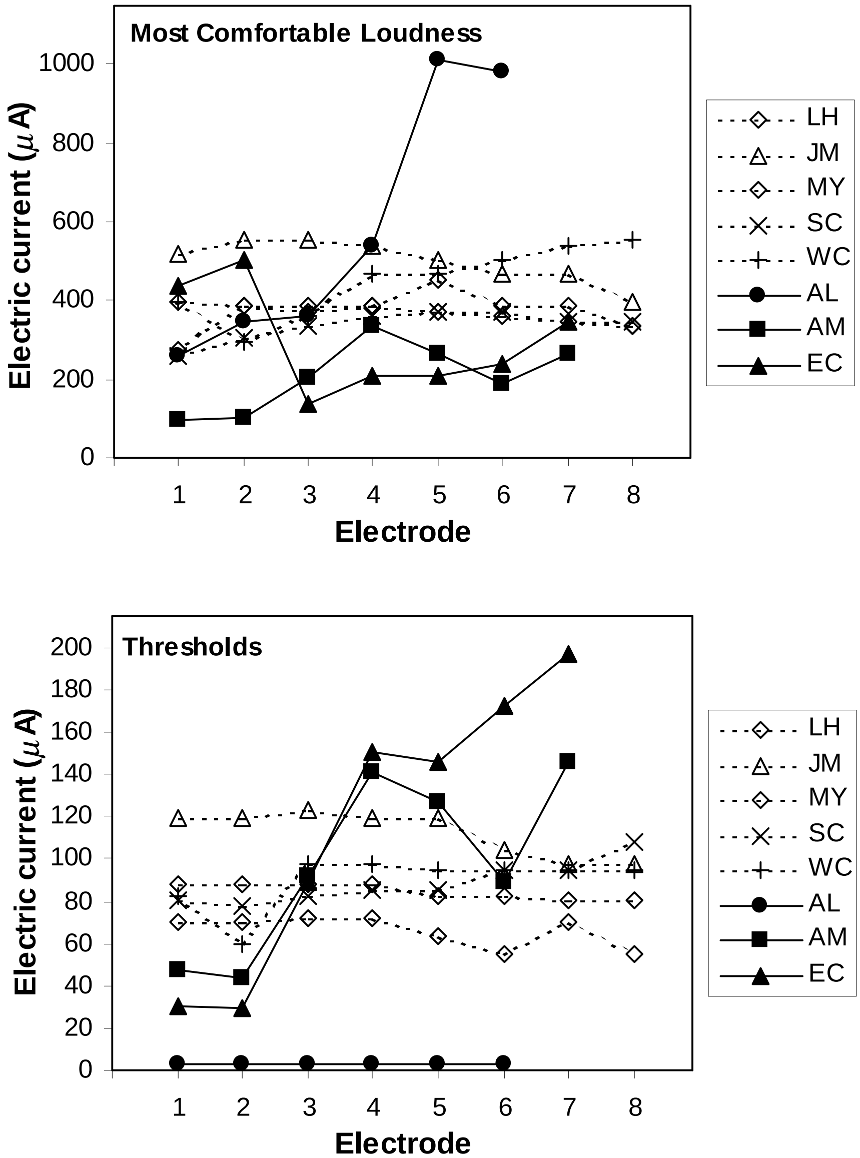
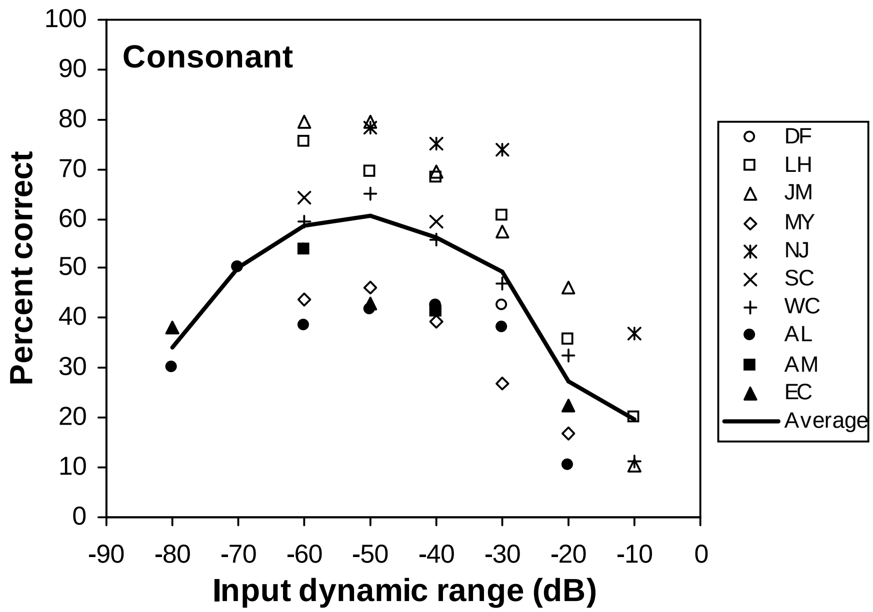


Figure 4. Consonant (top) and vowel (bottom) recognition scores (y-axis) in quiet as a function of the input dynamic range (x-axis). Individual data are represented by symbols (open symbols for CIS users and filled symbols for SAS users). The average data are represented by the solid line.



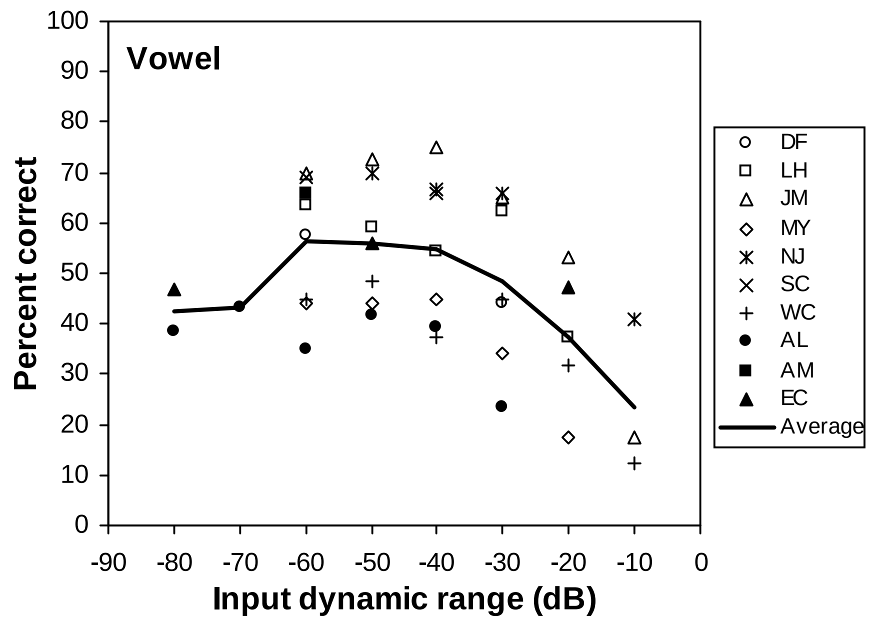
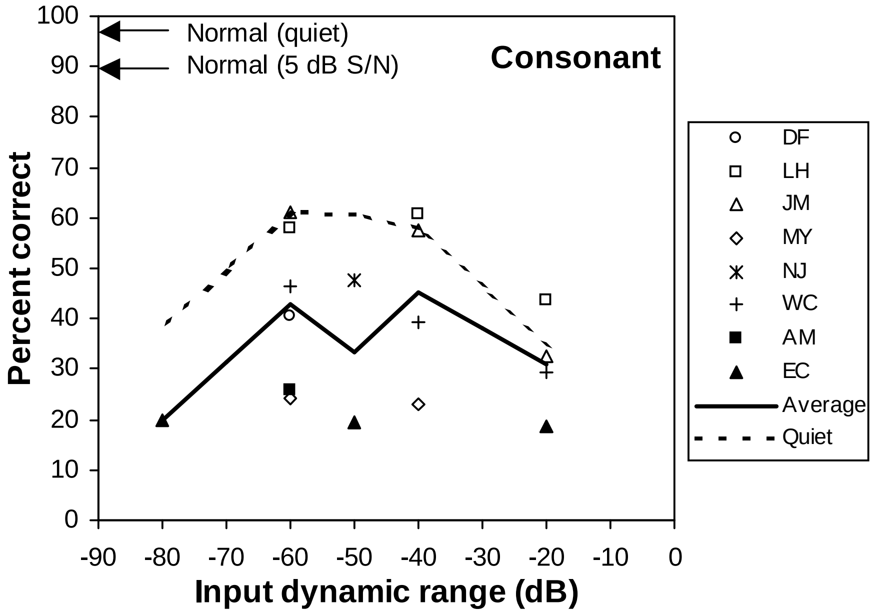


Figure 5. . Consonant (top) and vowel (bottom) recognition scores (y-axis) in noise as a function of the input dynamic range (x-axis). Individual data are represented by symbols (open symbols for CIS users and filled symbols for SAS users). The average data are represented by the solid line. For comparison, the correspondent average data in quiet (see text) are also included (dashed line).



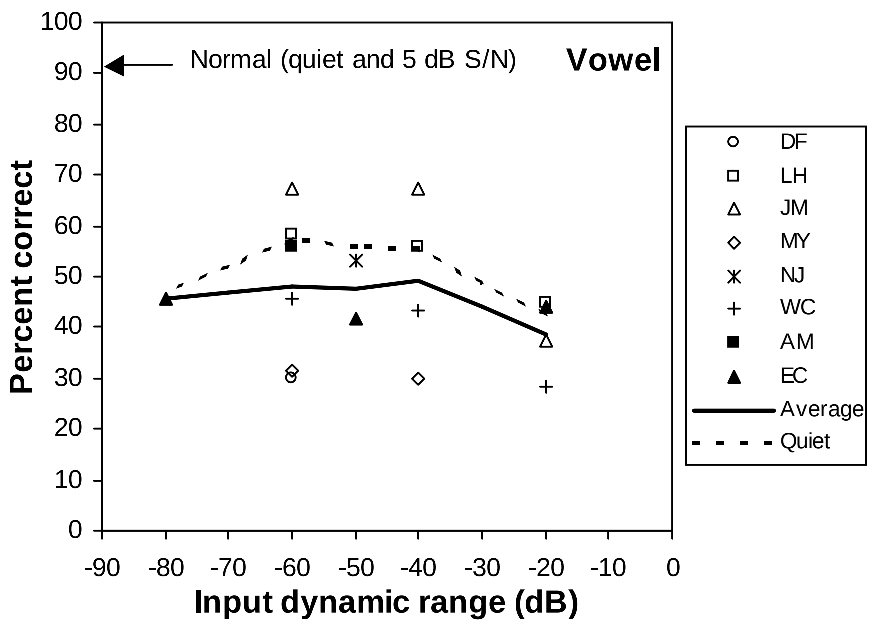


Figure 6. Effects of envelope level distribution. I. Logarithmic mapping for both consonants and vowels. The right-bottom panel shows idealized acoustic envelope level distribution for consonants (dotted line) and vowels (solid line). The right-top panel shows the logarithmic acoustic-to-electric conversion. The left-top panel shows electric envelope level distribution. Note that a significant portion of low electric envelope levels are mapped below threshold (T level) and also that a portion of electric dynamic range is unused (indicated by the line with arrow on both ends).

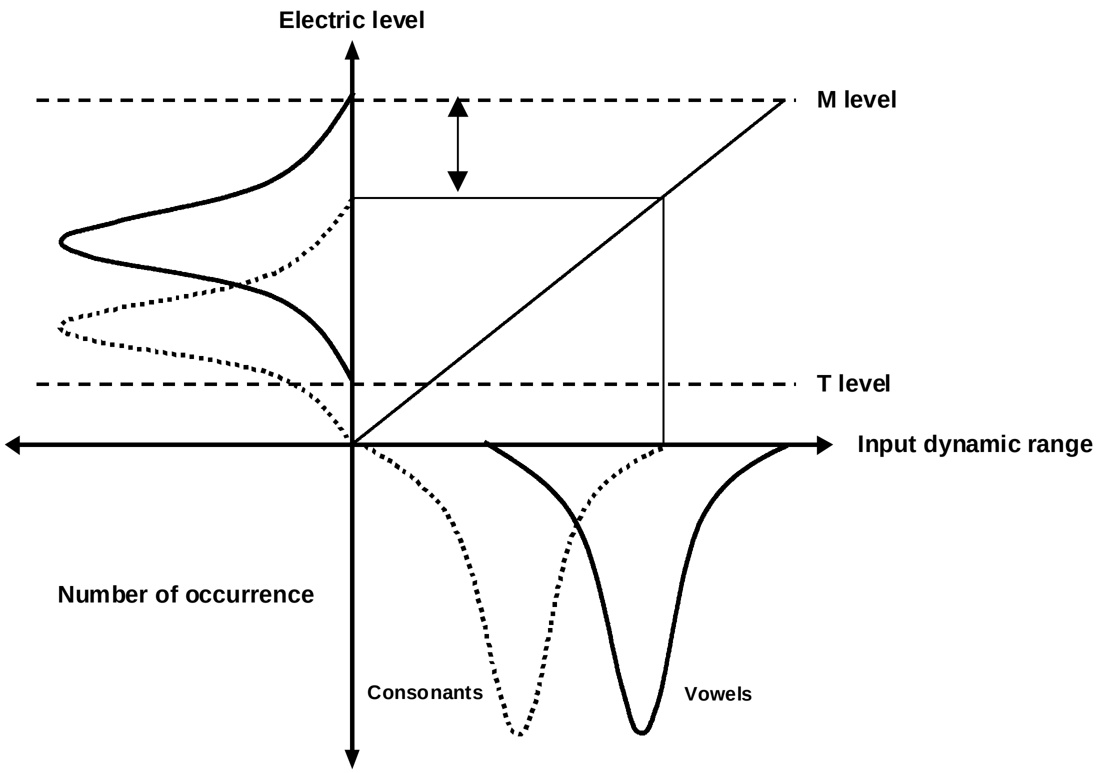
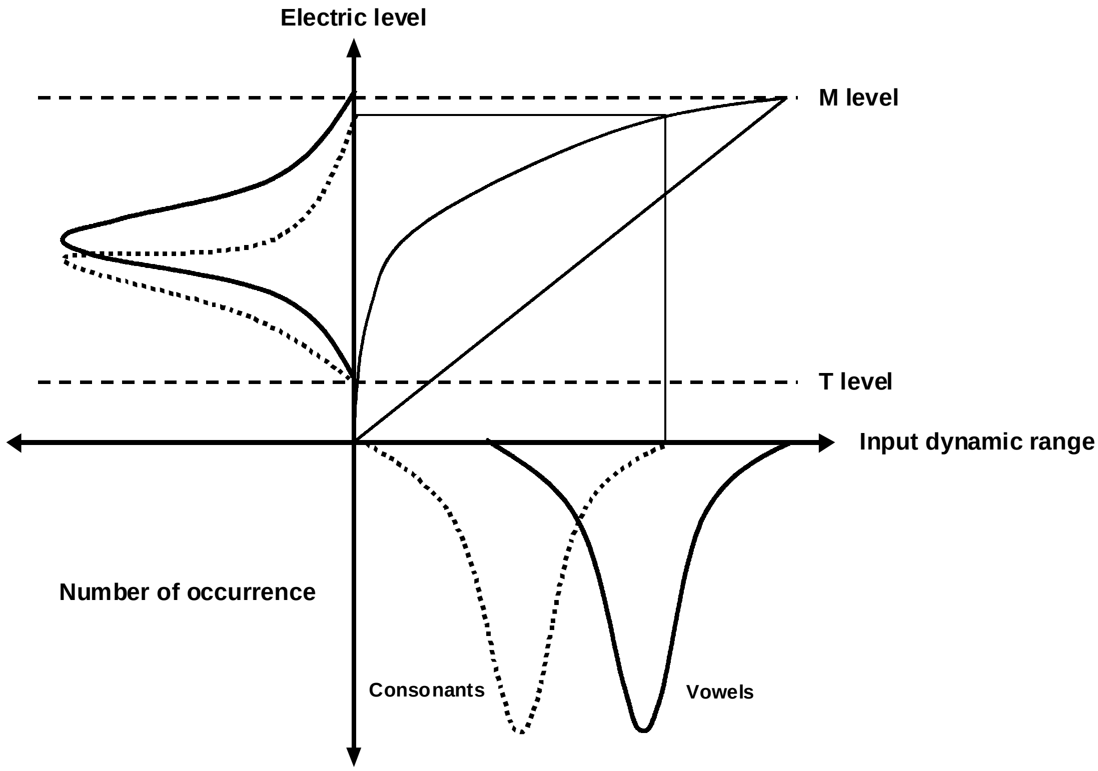


Figure 7. Effects of envelope level distribution. II. Logarithmic mapping for vowels and more compressive mapping for consonants. The right-bottom panel shows idealized acoustic envelope level distribution for consonants (dotted line) and vowels (solid line). The right-top panel shows the logarithmic acoustic-to-electric conversion (straight line) for vowels and the more compressive conversion (curved line) for consonants. The left-top panel shows electric envelope level distribution. Note the improved use of electric dynamic range for consonants.



Electric stimulation in auditory neuropathy (AN)

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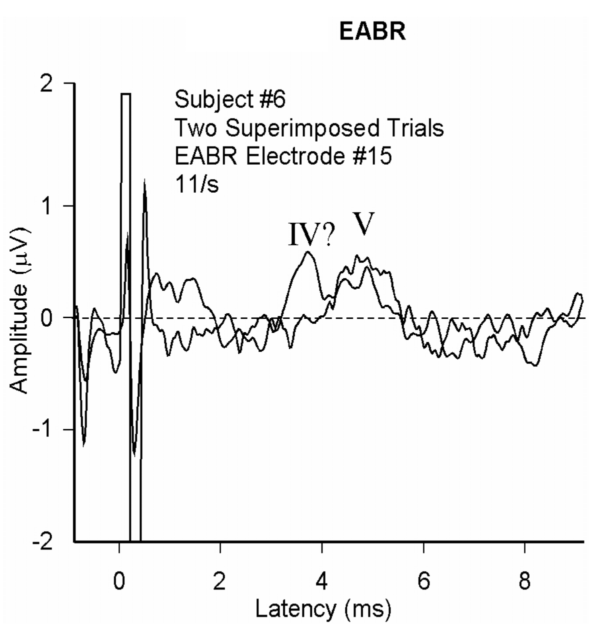
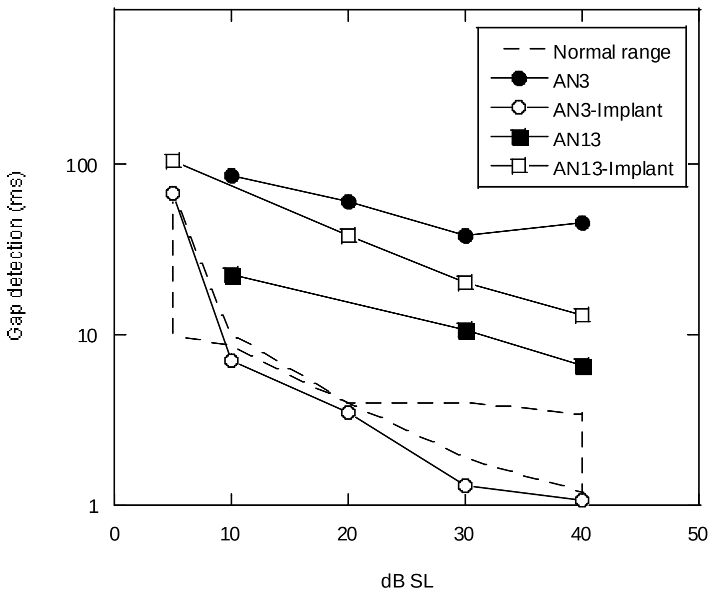
Recently there have been a large number of AN subjects (approximately 30 in US) who have received a cochlear implant. We have had the opportunity to study 2 AN subjects with cochlear implants in regard to the auditory pathway and psychoacoustic temporal processes. Some measures of auditory temporal processes showed improvement but the extent of improvement varied widely.

Results:
The first subject (AN3) had a profound hearing loss at the time of implantation with an associated peripheral neuropathy (Type I AN). Word comprehension was gradually lost over 15 years and was 0% at the time of implantation. ABRs were absent with acoustic stimulation while cochlear microphonics and TEOAEs were present. Gap detection to acoustic stimuli prior to implantation was profoundly elevated (80 ms).

We tested evoked potentials of the auditory pathway and psychophysical functions using electrical stimulation with the implant. For the electrical ABR (EABR), a brief electrical stimulus was presented at a rate of 11/s and with a current strength that elicited the sensation of a brief sound with a subjective “loudness” of “7” on a 1-10 scale (10 being very loud and 1 being very faint). Brain potentials (30-3000 Hz filter bandwidth) were averaged across 2000-3000 stimuli and EABRs were defined (see Figure below). The grand averaged EABR had a questionable component IV at 3.7ms (evident on only one the two averages) but a consistent component V at 4.9 ms. Component V was delayed approximately 1 ms compared to other studies of EABRs (e.g., Starr and Brackmann, 1979 reported Wave V at 3.9 ms; Brown et al., 2000). An averaged EABR was not elicited when the rate was increased to 24/s. This was in contrast to other non-AN subjects with cochlear sensory deafness who had EABRs with stimulus rates at 100/s. Thus, the EABR in this patient was abnormal, being limited to a Wave V that was delayed in latency at slow stimulus rates and absent when stimulus rates were increased, findings consistent with a neuropathic disorder of the auditory nerve.



Interestingly, this subject also had significant elevation of thresholds for discriminating changes of rate, requiring about 15 Hz to distinguish a change at 10 and 20 Hz (normal and implant users need about 5 Hz) and could not tell differences between stimulus rates above 50 Hz. In contrast, we found that electric stimulation totally restored normal gap detection to this neuropathy subject. Figure on the left shows normal-hearing listeners' gap detection threshold range (mean ± 2 SDs, shaded area); for this AN subject, the pre-surgical gap detection thresholds via acoustic stimulation (open circles) and post-surgical threshold via electric stimulation (filled circles). The data clearly show that the impaired gap detection threshold (80 ms vs. normal 2 ms at high sensation levels) was totally restored to the normal range (1.5 ms) with the cochlear implant.



On the other hand, the second neuropathy subject (AN13) had only moderate hearing loss prior to implantation, but could not recognize speech in moderately noisy backgrounds. She also had robust cochlear microphonics and otoacoustic emissions but no ABRs acoustic stimulation. With the cochlear implant (Nucleus 24), she still had a large gap detection threshold (see Figure above) of 13 msec even when the stimulus was presented at the maximum comfortable loudness level. Rate discrimination was also abnormal. She needed 24 and 26 Hz increase to tell there was a rate difference for the standard rate 10 and 20 Hz, respectively. She could not tell difference for rates above 200 Hz.

Because this subject (AN13) had almost normal audiogram on the non-implanted side, we compared consonant recognition with acoustic and electric stimulation in quiet (65 dBA) and in noise (S/N=+10 dB). Test stimuli and procedure were described in Progress Report #4. In quiet she achieved 68% consonant recognition with her non-implant ear (implant off) and 56% correct with her cochlear implant (normal ear plugged). However, in the presence of the noise, her score decreased to 29% with acoustic stimulation (typical result in AN subjects) but was relatively unchanged at 52% with the implant. More interesting results were found in the bilateral condition with both acoustic and electric stimulation. In quiet, she was able to integrate between ears to increase her recognition score to 82%. In noise, the score was 46%, which fell between the score for acoustic or electric stimulation alone.

Summary:

Our preliminary results showed that cochlear implants could improve auditory processing in AN subjects, but this improvement is not uniform. For example, AN subjects with cochlear implants could not process fast rate information. We need to measure similar temporal processing tests in additional AN subjects to determine whether poor rate discrimination is characteristic of all AN subjects. This information is important for optimizing implant stimulation procedures. Our preliminary data suggest that speech strategies with a low-rate, but a great number of channels of stimulation (e.g., 20-channel ACE strategy with low-rate stimulation) may be more advantageous than a fast-rate CIS strategy in AN subjects with cochlear implants.

Music perception in cochlear implant users

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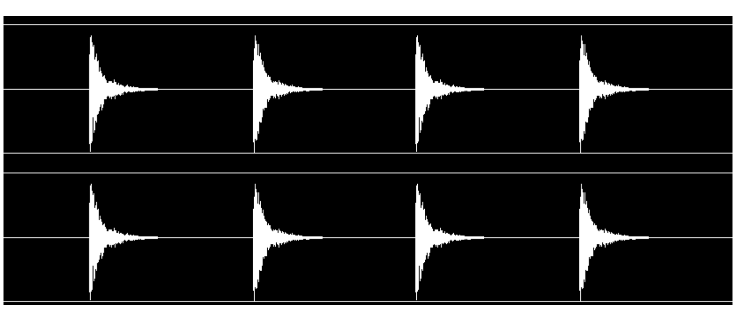
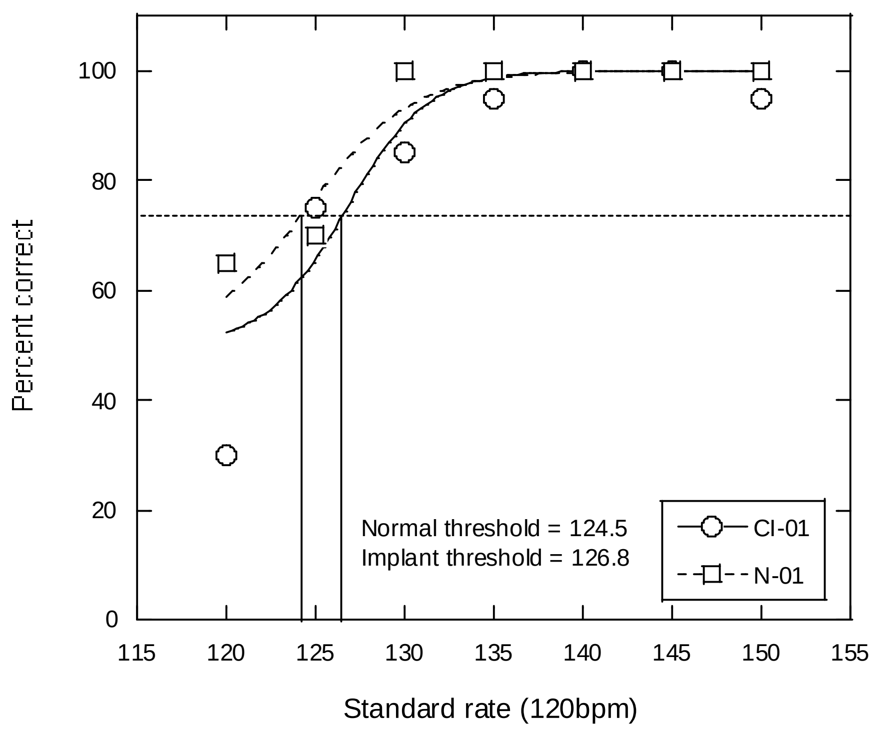
While many cochlear implant (CI) users enjoy success regarding speech understanding, most of them are still frustrated by their inability to accurately hear music. Also, CI users in non-western countries where languages are tonally based, do not seem to derive the same benefit as individuals who speak non-tonal languages. The overall objective of this study is to identify the critical information necessary for accurate perception of music and tonal languages. As part of a series of systematic studies, we present two experiments in tempo and rhythmic pattern discrimination.

Musical tempos and patterns were generated by an Alesis SR-16 drum machine. A kick drum sample and a snare drum sample were used to represent both low and mid-frequency information. Tempos ranged from 60 to 150 beats per minute (bpm). The tempo study used one bar of the same pattern, which varied by tempo each presentation. For the pattern discrimination study, six one-bar rhythmic patterns were used including permutations of quarter, eighth, and sixteenth notes. Figure below shows an audio file and the musical notation which corresponds to that sound pattern.




Both normal hearing and CI subjects listened to the stimuli in the sound field at a comfortable listening level (55-60 dBA). The CI users listened to the musical sounds through their speech processors using the normal setting. In the tempo discrimination experiment, the subject was required to identify the faster tempo in a two-interval forced-choice paradigm (2IFC) task. In the pattern discrimination experiment, the subject had to identify the rhythmic pattern that was different in a three-interval forced-choice (3IFC) task. A psychometric function was fitted to the data to derive a discrimination threshold for both experiments.

Preliminary data showed that there was no difference in tempo discrimination between normal hearing and CI listeners (see Figure below). However, as we continue to train our listeners on additional musical tasks we expect greater differences in their abilities to accurately hear music. For example, we found CI listeners could not identify commonly known melodies. These results suggest that cochlear implant listeners have relatively normal temporal processing but impaired processing of fine-frequency structure. To achieve a high level of musical appreciation, this fine-frequency structure has to be encoded in future cochlear implants.



Plans for the next quarter at UCI:

Hardware – Clarion Research Interface (CRI-II). We are in the process of obtaining the research interface for the new generation of Clarion cochlear implants (CRI-II). The new interface allows electric field measurement and many additional features that were not available in the previous devices. The UCI implant center has recently implanted 3 patients with the Clarion C-II device, and the House Ear Clinic also has 3 C-II patients. We will work with these patients to address electrode interaction in cochlear implants.

Experiments – Psychophysics. We will continue to collect intensity, temporal, and spectral processing data in cochlear implants. We hope these basic psychophysical data will form the basis for customized speech processing.

Experiments – Speech Processor Design. We will continue to recruit both good and poor cochlear implant users and hope to improve their performance in quiet and in noise. Specifically, we will evaluate whether (1) different amplitude mapping functions for different channels will produce improved speech recognition and (2) neuropathy subjects benefit from a speech strategy with a low rate but a high number of channels of stimulation.

Publications and Presentations in this Quarter:

Publications:

Chatterjee, M., Shannon. R.V., Galvin, J.J. and Fu, Q.-J. (2001). Spread of excitation and its influence on auditory perception with cochlear implants, Physiological and Psychological bases of Auditory Function: Proceedings of the 12th International Symposium on Hearing, A.J.M. Houtsma, A. Kohlrausch, V.F. Prijs, and R. Schoonhoven (Eds.), Shaker Publishing BV, Maastricht, NL, pp. 403-410.

Fu, Q.-J., Galvin, J., and Wang, X. (2001). Recognition of time-distorted sentences by normal-hearing and cochlear-implant listeners, J. Acoust. Soc. Amer, 109(1), 379-384.

Fu, Q-J. and Galvin, J. (2001). Recognition of spectrally asynchronous speech by normal-hearing listeners and Nucleus-22 cochlear implant users, J. Acoust. Soc. Amer., 109(3), 1166-1172.

Fu, Q.-J., and Zeng, F.-G. (2000). Identification of temporal envelope cues in Chinese tone recognition. Asia Pacific Journal of Speech, Language and Hearing 5, 45-57.

Shannon, R.V. Fu, Q.-J., Wang, X., Galvin, J. and Wygonski, J. (2001). Critical cues for auditory pattern recognition in speech: Implications for cochlear implant speech processor design, Physiological and Psychological bases of Auditory Function: Proceedings of the 12th International Symposium on Hearing, A.J.M. Houtsma, A. Kohlrausch, V.F. Prijs, and R. Schoonhoven (Eds.), Shaker Publishing BV, Maastricht, NL, pp. 500-508.

Shannon, R.V. (2001). Have you hugged your modiolus today, CIAI Contact, winter.

Shannon, R.V. (2001). Implants in the brain, CIAI Contact, winter.

Shannon, R.V. (2001). Overview of cochlear implant technology and research, in proceedings of “Nonsyndromic deafness: Clinical issues and research opportunities”, Short Course at the 2001 Midwinter Meeting of the Association for Research in Otolaryngology, St. Petersburg Beach, FL., Feb 4-8.

Zeng, F.-G., Fu, Q.-J., and Morse, R.P. (2000). Human hearing enhanced by noise. Brain Research 869(1-2), 251-255.

Zeng, F.-G. (2000). Advances in cochlear implants. In: Speech and Hearing Review - A Bilingual Annual Publication, pp. 6-17 (Chinese) and pp. 54-61 (English). L.-R. L. Chen (Ed.), National Kaohsiung Normal University Special Education Center, Taiwan.

Zeng, F.-G. (2000). Auditory Neuropathy: Why some hearing-impaired listeners can hear but do not understand and how can DSP technology help them? IEEE Signal Processing Society - Ninth DSP (DSP 2000) Workshop, October 15-18, 2000, Hunt, Texas.

Zeng, F.-G., Oba, S., and Starr, A. (2001). Suprathreshold processing deficits due to desynchronous neural activities in Auditory Neuropathy. In: Physiological and Psychophysical Bases of Auditory Function, pp. 365-

372. D.J. Breebaart, Houstma, A.J.M., Kohlrausch, A., Prijs, V.F., & Schoonhoven, R. (Eds.), Shaker publishing BV, Maastricht, Netherlands.

Presenations:

Chatterjee, M. and Galvin, J. (2001). Modulation masking in cochlear implant listeners. 2001 Midwinter Meeting of the Association for Research in Otolaryngology, St. Petersburg Beach, FL., Feb 4-8. (poster)

Friesen,L. and Shannon, R.V. (2001). Speech recognition as a function of stimulation rate in Clarion and Nucleus-22 cochlear implants, 2001 Midwinter Meeting of the Association for Research in Otolaryngology, St. Petersburg Beach, FL., Feb 4-8. (poster)

Friesen,L. and Shannon, R.V. (2001). The effect of stimulation rate on speech recognition in Clarion and Nucleus-24 cochlear implant listeners, 8th Symposium on Cochlear Implants in Children, Los Angeles, CA Feb 28 – March 3. (poster)

Fu, Q.-J. (2001). High variability of speech performance in cochlear implant users: Contribution of auditory resolution, 2001 Midwinter Meeting of the Association for Research in Otolaryngology, St. Petersburg Beach, FL., Feb 4-8. (poster)

Kwon, B.J. and Shannon, R.V. (2001). Is modulation masking due to a modulation filterbank or simply temporal interference?, 2001 Midwinter Meeting of the Association for Research in Otolaryngology, St. Petersburg Beach, FL., Feb 4-8. (poster)

Wei, C.G., Zeng, F.-G., Cao, K.L., and Wang, Z.Z. (2001). Rate discrimination and tone recognition in Chinese cochlear-implant listeners. Abstracts of the 24th ARO Midwinter Research Meeting, P.A. Santi (Ed.), 293, St. Petersburg, Florida.

Zeng, F.-G., Starr, A., and Oba, S. (2000). Auditory Neuropathy: “ I can hear but do not understand,” National Academy of Science Colloquium, 19-21 May 2000, Beckman Center, Irvine, California.

Zeng, F.-G. (2000). The upper boundary of temporal pitch coding inferred from electric stimulation of the auditory nerve. The Journal of Acoustical Society of America 108, 5(2), 2596, Joint 140th Meeting ASA and Noise Control 2000, Newport Beach, California.

Invited Presentations:

Shannon, R.V. (2001). Cochlear implants in young children: Ethical issues in a rapidly changing technology, Department of Psychology, UC Berkeley, Jan 22.

Shannon, R.V. and Baskent, D. (2001). Speech recognition under conditions of frequency-place expansion and compression, Dept. Of Otolaryngology, UCSF, Jan 23.

Shannon, R.V. (2001). Cochlear implants: Ethical issues in a rapidly changing technology, 2001 Midwinter Meeting of the Association for Research in Otolaryngology, St. Petersburg Beach, FL., Feb 4-8.

Shannon, R.V. (2001). Overview of cochlear implant technology and research, lecture in the short course “Nonsyndromic deafness: Clinical issues and research opportunities, 2001 Midwinter Meeting of the Association for Research in Otolaryngology, St. Petersburg Beach, FL., Feb 4-8.

Shannon, R.V. (2001). The Penetrating electrode ABI, at Cochlear Corp. Symposium “New Frontiers in Implant Outcomes and Technology”, Los Angeles, Feb 28.

Shannon, R.V. (2001). Basic anatomy of the cerebello-pontine angle region, ABI Training Course, Los Angeles, March 4.

Shannon, R.V. and Baskent, D. (2001). Speech recognition under conditions of frequency-place expansion and compression, Dept. of Electrical Engineering, University of Michigan, March 20-21.

Zeng, F.-G. (2000). Auditory information processing: What have we learned from cochlear implants? Department of Biomedical Engineering, Johns Hopkins University (5/12). Live webcast: http://waverly.bme.jhu.edu/ramgen/20000512.rm

Zeng, F.-G. (2000). Auditory Neuropathy: Why some hearing-impaired listeners can hear but cannot understand speech? The First Annual Research Symposium, Department of Otolaryngology, University of California, Irvine.

Zeng, F.-G. (2000). What have cochlear implants told us about basis issues in hearing? Department of Otolaryngology, University of California, San Francisco.

Zeng, F.-G. (2000). What have cochlear implants told us about basis issues in hearing? Department of Psychology, University of California, Berkeley.

Zeng, F.-G. (2001). Speech recognition via bionic ears. A Special Presentation for UCI Health Science Partners and Board of Visitors, Beckman Center of the National Academies, Irvine, California.

Zeng, F.-G., Stickney, G., and Cruz, R. (2001). Advances in cochlear implants. Oralingua School for the Hearing Impaired, Whittier, California.

Zeng, F.-G. (2001). What have cochlear implants told us about basis issues in hearing? Kresge Hearing Research Institute, Department of Otolaryngology, University of Michigan, Ann Arbor.

Zeng, F.-G. (2001). Intensity coding in acoustic and electric hearing. Department of Cognitive Sciences, University of California, Irvine.