occasions from active electrode 1 differed less than 5%, which is within the range of variability established earlier,\(^1\) despite a considerable decrease in \(C\) levels between the two occasions. It was concluded that the device performed normally and that a fluctuation in responsiveness was the most likely cause. Patient 55, who was either congenitally deaf or deafened at age 1 by meningitis, was implanted at the age of 9. Device fitting was difficult, because \(T\) and \(C\) levels in the common ground mode appeared to change drastically within one session. Overall, \(T\) and \(C\) levels increased drastically within the initial 5 months. In this case the AEVs from active electrode 1 at a constant current level decreased 24%, suggesting encapsulation of the implanted electrodes by new bone. Tissue growth at the round window is a less likely explanation for the decrease of the AEVs, as it would not affect response thresholds.

In our population 3 adult patients were deaf owing to otosclerosis. These cases showed an E-E map with massive phase reversal, which suggests that the current generating surface potentials flow through very permeable cochlear bone.\(^4\) In contrast, the \(T\) and \(C\) levels shown in Fig. 4 appeared normal in 2 tested patients, even for the "pseudomonopolar" mode (the most basal reference electrode). Pitch scaling in the pseudomonopolar mode, in which the patient had to rank a 300-millisecond tone burst on a six-point scale from "very high" to "very low," was performed. All electrodes were tested three times in random order. A normal tonotopy was found in patient 19 (as expected from the literature\(^5\)), but in patient 58 normal tonotopy was disturbed by a complex tone perceived when apical electrodes 18 to 20 were stimulated. So, only in 1 out of 2 patients deafened by otosclerosis, and only while stimulating across the basal turn of the cochlea, was the effect of the abnormal current flow found, and the effect was only on pitch perception and not on the threshold. Apparently, the extracochlear current flow picked up in these measurements does not reveal the (radial) stimulating current flow. However, the present data do seem to pose an interesting limiting case for field models of the electrically stimulated ear. They also underline that deviant AEVs from abnormal cochleas should not be interpreted too easily as caused by faulty electrode placement or by an electrode failure.

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REFERENCES


ADJUSTMENT OF APPROPRIATE SIGNAL LEVELS IN THE SPECTRA 22 AND MINI SPEECH PROCESSORS

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INTRODUCTION

The Spectra 22 speech processor has been described (Seligman and McDermott, this suppl, section 6). Figure 1 shows the audio signal path and means of mapping loudness in this processor and its predecessor, the Cochlear Pty Limited Mini Speech Processor (MSP). In both processors, following the microphone and preamplification, the signal level is adjusted by a sensitivity control. This control is the equivalent of the input gain of a hearing aid and is quite distinct from a loudness or maximum output level control. As will be explained later in this paper, the setting of the control is crucial to the effective functioning of the speech processor.

An automatic gain control (AGC) follows the sensitivity control. The AGC has the function of preventing limiting or saturation in the remaining signal processing. In the Cochlear speech processors this is of the "infinite compression" type. In this form of AGC, the amplifier is linear, but at a certain point, starts to reduce the gain so that any increase in input level is reduced to prevent the output level from rising further. The attack time is about 1 millisecond (ms) and the decay time 50 ms.

Following the AGC, the signal is analyzed by the signal processor. This analysis consists of filtering by a filter bank in the Spectra 22, or feature extraction and filtering in the MSP. In this discussion, the type of signal processing is not important.

Fig. 1. Speech processor audio signal path. AGC — automatic gain control, \(T\) & \(C\) — threshold and comfortable loudness.
Output Level Range

Fig 2. Loudness mapping function. At correct sensitivity setting, input level of 150 corresponds to SPL of 74 dB. Vertical axis represents log charge delivered between patients’ own specific threshold (T) and comfortable loudness (C) levels.

Data are available from the signal processor in 8-bit digital form. At this stage the amplitudes of filter bank (or feature extractor) outputs are converted via a lookup table to 8-bit decibel value. This table (Fig 2), in combination with the patient’s measured threshold (T) and comfortable loudness (C) levels, determines the level of the electrical stimulus. The function has been selected to give optimal speech performance for the majority of patients. While the maximum input level is 255, to give some headroom, 150 gives C level stimulation. The lower end of this function is the so-called base level, which is the level below which no stimulation occurs. This is typically set to 4 and the range between first stimulation, S, and 150 is 30 dB. The relationship between the C level at an input of 150 and the sound pressure level (SPL) depends on the setting of the sensitivity control.

IMPORTANCE OF CORRECT SENSITIVITY SETTING

From an engineering point of view, if there is an AGC before the signal processor, everything is under control. However, the practicalities of the matter for a patient are quite different. Consider a cocktail party situation, with the sensitivity control set so that the AGC is just operating at peaks of 74 dB SPL. This makes speech discrimination for the cochlear implant recipient extremely difficult.

Now consider the case in which the sensitivity control has been set too low. The dynamic range of the signal is restricted because the maximum level is reduced but the base level is fixed. This is depicted in Fig 3B.

It may be asked, in view of the desirability of setting the sensitivity control so that the AGC is mostly not operating, why have an AGC at all? The reason is that one cannot ensure that the signal will not be cut off. When the patients themselves speak, the signal level at their own microphones, only 15 cm from the mouth, is quite different than the signal level of the people to whom they are speaking, who may be several meters away. Of course, signal levels also vary considerably between speakers and within individual speakers. It is the function of the AGC to cope with the individual differences in signal level in a given situation, rather than to adapt to different situations.

SETTING OF OPTIMAL SENSITIVITY

To assist in the appropriate setting of an optimal sensitivity, the Cochlear Diagnostic and Programming System (DPS) provides a software function and procedure. This is essentially a means of displaying visually, on an array of lights, the setting of the AGC gain. The lights normally provide a display of stimulated electrodes, but their role here is changed. By using this function, the audiologist can find the correct setting, first using a fixed-level input signal provided by the programming hardware, and then using live voice. The object is to set the sensitivity so that the AGC is just on the point of operating at peaks of 74 dB SPL. By means of this sensitivity adjustment program, adjustments can be made for live and recorded material presented either through a loudspeaker or directly into the external input socket of the speech processor. It can

background noise level. When the person to whom the patient is talking speaks, the compression action of the AGC prevents the level from getting any higher. This makes speech discrimination for the cochlear implant recipient extremely difficult. There is no level difference between the signal and the noise. So high a sensitivity setting reduces the apparent dynamic range, because the AGC reduces the signal while amplifying the noise in the gaps. The resulting reduced dynamic range is shown in Fig 3A.

Fig 3. Effect of sensitivity setting. A) Too high. B) Too low.
be seen that speech testing without a rigid protocol for the adjustment of signal level is hazardous. To allow the patient to adjust the sensitivity at each test session introduces a variable that may in some circumstances overwhelm the difference between the test conditions.

SENSITIVITY SETTING IN CLINICAL SITUATION

It is worth examining some reasons for inappropriate sensitivity settings in practice. If under normal conditions, the patient sets the sensitivity higher than the optimal setting, the most likely cause is that the C levels are too low. In an attempt to raise the loudness, the patient has set the sensitivity so that C levels are more frequently reached. A similar effect can be produced through the T levels' being set below threshold. The quieter sounds may be inaudible and the patient, in an attempt to bring these up, sets the sensitivity high. This produces the reduced dynamic range situation shown in Fig 3A.

The converse of the above is the patient who wears the processor set to a very low sensitivity. Here it is likely that either the C or T levels are set too high. To avoid unpleasantly loud sounds, the patient turns the sensitivity down. If the C levels are too high, loud sounds will be presented at an uncomfortable level. If the T levels are too high, any background noise above the base level will be loud and annoying. The result is the reduced dynamic range situation in Fig 3B.

AUTOMATIC SENSITIVITY CONTROL

If the T and C levels and sensitivity are all correctly set, there obviously remain the highly variable acoustic environment conditions, comprising both signal and background noise levels. Normal-hearing people, of course, do not need to adjust their sensitivity controls as conditions change. In the Spectra 22 and MSP, a function is provided to attempt to automate the adjustment. This is selected by choosing a setting marked "S." On this setting, the processor carries out an adjustment based on the noise floor of the sound. The noise floor is the level to which the sound drops during breaks in speech. If one were trying to measure background noise level while someone was speaking, a way to do it (apart from telling the person to stop) would be to look at the lowest level the sound reaches in the breaks in the speech. The speech processor does the same. On the S setting it monitors the noise floor level continuously via a minimum level detector. If the noise floor is above a programmable "break point," it is likely that the gain is too high for that situation. Here the automatic sensitivity control gradually reduces the sensitivity. This is done via an electronic attenuator that follows the manual control as shown in Fig 4.

The effect of the automatic sensitivity control is illustrated in Fig 5. The top trace in Fig 5A shows the amplitude of the utterance "They looked up at the blue sky" spoken in the presence of eight-talker babble. In the lower trace it can be seen that the AGC gain reduces significantly during the words of the utterance but goes to maximum during the breaks. This reduces the amplitude difference between the signal and the babble. In Fig 5B, the S function is switched on. In this case the automatic sensitivity control has reduced the gain. The lower trace shows that the AGC is sitting on maximum gain.
nearly all the time and only reduces on some peaks. The much-reduced noise floor can be seen in the upper trace. (It will be noticed that in Fig 5B, the noise floor has been almost completely removed. Obviously this is not something that can be done simply by reducing the gain, which can only prevent the adverse effect of the AGC. However, in the MSP, the remaining noise floor can be subtracted from the formant amplitude, giving the result shown. In the Spectra 22, because there are no formant amplitudes as such, this technique is not available. The output of each individual filter does not have a consistent noise floor. Therefore, an alternative method is used: on the S setting the base level is increased by six levels.)

CONCLUSIONS
1. In spite of the use of an AGC, there is an "optimal" sensitivity setting that maximizes the perceived dynamic range presented to a cochlear implant patient.
2. The optimal sensitivity setting maximizes the instantaneous dynamic range presented.
3. Incorrect settings of the sensitivity control may impinge on patient speech performance.
4. Incorrect T and/or C level settings may result in attempts by the patient to compensate for this by inappropriate use of the sensitivity control.
5. A means is available in the Spectra 22 and MSP to automatically adjust the sensitivity control in noisy situations.

RELATIONSHIPS AMONG COMFORT LEVELS DETERMINED BY COCHLEAR IMPLANT PATIENT’S SELF-PROGRAMMING, AUDIOLOGIST’S PROGRAMMING, AND ELECTRICAL STAPEDIUS REFLEX THRESHOLDS

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INTRODUCTION
The electrical stapedius reflex (ESR) threshold has been utilized in the prediction of comfort levels for cochlear implant patients.1-5 The previous studies have suggested that the ESR threshold correlates significantly with behavioral comfort level. Typically, the method utilized to assess comfort levels is either controlled by the audiologist, using the keyboard (method of limits), or by the patient, using an adjustable control knob (method of adjustment). We have used various psychophysical methods in programming our adult patients and decided to determine if these methods would result in significant differences for the measurement of most comfortable levels (MCLs) and uncomfortable levels (UCLs). Finally, we wanted to know if the ESR threshold would correlate best with MCL or UCL.

SUBJECTS AND METHODS
Twenty-two adult cochlear implant patients who had used their Nucleus multichannel cochlear implant for at least 1 year participated in this research. The subjects were selected at random from our population of 155 adult multichannel cochlear implant patients. The mean age of the subjects was 44.9 years and the mean length of implant use was 48.3 months. Eight of the subjects were female and the remaining 14 subjects were male. The MCL, UCL, and ESR measures were all obtained in stimulus level in the patient’s program mode, bipolar (N = 2), bipolar + 1 (N = 13), and bipolar + 2 (N = 2). Each subject was first screened for the presence of an ESR. This screening eliminated 5 subjects who did not have an ESR on any of the six electrodes we used in our protocol. The MCLs and UCLs were obtained from the remaining 17 subjects (7 female and 10 male) on six electrodes (two basal, two medial, and two apical) by three methods. These methods were 1) audiologist programming by method of limits with keyboard (method A), 2) patient self-programming by method of limits with keyboard (method S), and 3) patient self-pro-

Fig 1. Most comfortable (MCL) and uncomfortable (UCL) levels set by three methods. All measures are in programming units. A) Apical electrodes (N = 17). B) Basal electrodes (N = 17).
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