the quasi-static approximation used in this letter is not likely to be valid for the case of objects whose dimensions are comparable to or larger than about $\lambda/15$ or $\lambda/10$, where λ is the free-space wavelength. A more detailed calculation of the field lines terminating on such metallic structures would be necessary to estimate I_{sc} and hence I_h.

In order to confirm some of the highlights of the above calculations, experiments have been performed with reduced-scale toy models of a car and a jeep using a 0.1 λ monopole-above-ground antenna to which power was fed at 39 MHz with a coupled transformer balun. An average I_{sc}/E of 1.17 (mA)/(V/m) measured for the full-scale version of the toy-model car (scaling factor = 51) is in good agreement with the value of 1.12 (mA)/(V/m) calculated for the compact car identified in Table I for the simulated frequency of 39/51 or 0.765 MHz. Similarly believable results are also obtained from the model of the jeep.

Conclusions

A simple analysis based on the equivalent circuit representation of a human in conductive contact with an ungrounded, metallic object in a quasi-static HF field points out that there may be situations where the thresholds of perception and let-go can be exceeded for fields considerably lower than the ANSI recommended guideline of 615 V/m, the far-field equivalent E-field associated with a power density of 100 mW/cm^2 in the frequency band 0.3 to 3.0 MHz.

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A PARCOR Characterization of the Ear for Hearing Aids

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***Abstract*—A possible characterization of the ear through the PARCOR algorithm is suggested. This leads to a digital filter hearing aid that could be of assistance in compensating for partial hearing loss.**

I. The PARCOR Algorithm as an Incremental Model

The PARCOR algorithm [1, pp. 32-41] is an algorithm for linear prediction [2] which places considerable emphasis on reflection coefficient relationships of an acoustic-wave transmission system [3]-[5]. As such, it has been successfully used in several fields, including voice analysis [6] and synthesis [7] and geophysical signal deconvolution [8, pp. 125-135]. One of the PARCOR algorithm's major features is its incremental nature which gives it great modeling capability. In fact, the first uses of the algorithm have been in the uniformization of nonuniform structures, such as the vocal tract, involved in problems of wave propagation [3]-[7].

It seems, therefore, that a generalization in the use of the algorithm can be made to those problems necessitating numerical solution for traveling waves in nonuniform nonhomogeneous structures [9]. In this way, we are led to subdivide nonuniform systems into incremental parcels characterized by reflection (and transmission) coefficients from which numerical solutions are obtained. By this method the propagating medium becomes characterized by a concatenation of a finite number of uniform parcels described by their transmission and reflection parameters. The propagating medium which is originally described by continuous functions of space now becomes described by a finite number of parameters related with a spatial discretization.

One such nonuniform, nonhomogeneous, structure of great interest is the ear. In regard to the ear, solution of the sound propagation equations can give important information about many kinds of deafness, especially those resulting from mechanical transmission difficulties. In view of the above it would seem possible to apply the PARCOR algorithm to characterize with some success the different transmission patterns of damaged ears and thereby characterize different types and grades of deafness.

II. Acoustical Structure and Models of the Ear

As a processor of sound waves the ear can be considered to be an acoustical structure. Although the outer ear is simply described as a wave-transmitting structure [10, p. 77], the middle ear is more complicated [10, pp. 78-80] but still amenable, and similarly for the inner ear [10, pp. 81-83]. It also appears to be true for mechanical-to-neural transduction [10, pp. 83-86] and neural pathways to the auditory cortex [10, pp. 86-91]. It appears, that to some of these portions of the ear, the PARCOR algorithm can be applied, while this is especially of interest for the middle and inner ear where many damages occur causing hearing loss [11, pp. 3-27] due to aging, illness, and accidents. Application to the neural-related portions is more speculative but appears to be possible while each of the other problems can cause modifications to the bony chain responsible for mechanical wave transmission. This bony chain can be modeled by rigid bars with nonuniform cross section and variable elasticity which can then be broken into incremental parcels characterized by reflection–transmission coefficients for PARCOR algorithm use.

There is the possibility of extracting the necessary information through deconvolving procedures as used in geophysical analysis of reflections [8]. For this, one can use an acoustic stimulation of the mechanical–acoustical system formed by the outer–middle ear while monitoring reflections in the outer ear [12]-[14]. These reflections can be deconvolved by the PARCOR algorithm to give values of the reflection coefficients associated with the bony chain [15]. For the inner ear, the main element, the cochlea, has been modeled by various wave-transmission means [16]-[18] some of which [10, pp. 103-106] are directly applicable to the PARCOR algorithm. Indeed, the Zwislocki's model [19] lends itself to a good approximation of the cochlea through the partition membrane's movement and thus should allow an easy uniformization by a PARCOR model.

Due to the interconnection of the cochlea and the bony chain, it should be possible for a significant percentage of cochlear reflections to be sent back to the outer-ear canal, along with reflections from the middle ear. Thus the deconvolution algorithm should be able to obtain the necessary information about the mechanical characteristics of the cochlea, cochlear liquids, and partition membranes. In this way, the incremental analysis procedure through the PARCOR algorithm is seen as a possibly very useful tool for characterization of the mechanical transmission in the outer, middle, and inner sections of the ear.

III. A Possible PARCOR Algorithm Hearing Aid

By using measurements as suggested in [15] and discussed below, and possibly others, we can suppose that we are able to apply the PARCOR algorithm to calculate a set of reflection coefficients for a given ear for sections between the outer ear and the mechanical-to-neural tranducers. This can be done twice, once for a given deaf ear and once to an (averaged) normal ear yielding two all-pole (digital) transfer functions, $T_D(z)$ and $T_N(z)$, respectively, where z^{-1} is the unit delay. Assuming the same number of poles, M, in each, we can write for these transfer functions [6, p. 400]

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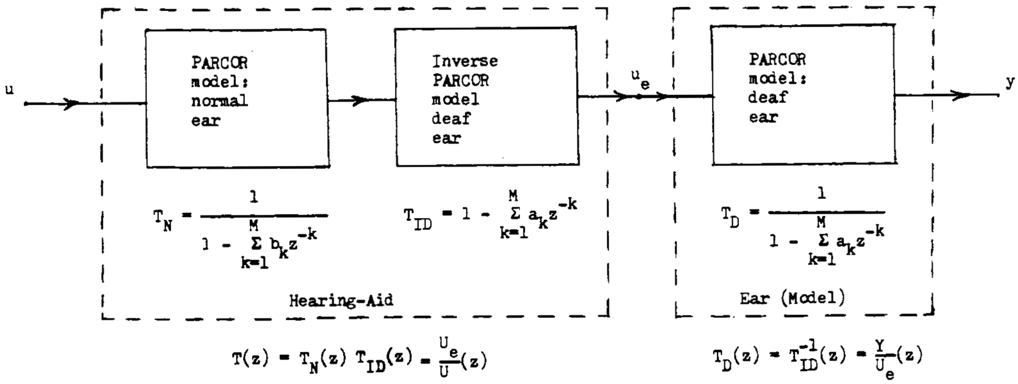

u = true sound signal

u_e = sound signal into deaf ear

y = representation of nerve sound signals to brain

Fig. 1. Hearing aid–ear digital filter structure.

$$T_D(z) = \frac{1}{1 - \sum_{k=1}^{M} a_k z^{-k}} \tag{1}$$

$$T_N(z) = \frac{1}{1 - \sum_{k=1}^{M} b_k z^{-k}}. \tag{2}$$

Consequently, if we preceed the deaf ear by the inverse of T_D

$$T_{ID} = T_D^{-1} = 1 - \sum_{k=1}^{M} a_k z^{-k} \tag{3}$$

we, in essence, cancel out the effects of the conductive component of the deafness. Preceeding T_{ID} by T_N brings the signal, coming out of the ear and into the transduction system, back to what would be present in the normal ear. As illustrated in Fig. 1, the hearing aid is the cascade of the realizations of T_N and T_{ID}. Thus the hearing aid is the digital filter

$$T(z) = T_N \cdot T_{ID} = \frac{1 - \sum_{k=1}^{M} a_k z^{-k}}{1 - \sum_{k=1}^{M} b_k z^{-k}} \tag{4}$$

with appropriate A/D and D/A conversion at the input and output ports. It is clear that such a device could easily be implemented on a VLSI chip suitable for mounting in the ear and that great adaptability of the device is possible.

Using the PARCOR algorithm we can give explicit formulas for the filter coefficients as follows. Suppose that $u_0^+(i)$ is the acoustic excitation wave value at the ear-canal entrance sampled at time t_i, $i = 0, 1, \cdots, m$. Let $u_0^-(i)$ be the response wave value at the same time and place (that is, the Kemp echo). This allows calculation of the reflection coefficient ρ_0 via the expression (set $k = 0$) [20, p. 424]

$$\rho_k = \frac{\sum_{i=1}^{M+m} u_k^-(i)\, u_k^+(i-1)}{\left[\left(\sum_{i=1}^{M+m} u_k^-(i)^2\right)\left(\sum_{i=1}^{M+m} u_k^+(i-1)^2\right)\right]^{1/2}}, \qquad k = 0, 1, \cdots, M. \tag{5}$$

In (5), m is the width of the sampling window and M is the degree of the digital filter. Next we determine the remaining ρ_k recursively using (5) for which $u_k^+(i)$ and $u_k^-(i)$, $k = 1, \cdots, M$, are found from [where $u_k^+(-1) = 0$]

$$\begin{bmatrix} u_{k+1}^+(i) \\ u_{k+1}^-(i) \end{bmatrix} = \frac{1}{1+\rho_k} \begin{bmatrix} 1 & -\rho_k \\ -\rho_k & 1 \end{bmatrix} \begin{bmatrix} u_k^+(i-1) \\ u_k^-(i) \end{bmatrix} \tag{6}$$

Supposing that we have calculated the reflection coefficients, ρ_k, $k = 0, 1, \cdots, M$, for the deaf ear, then the filter coefficients a_k are calculated recursively as follows [6, p. 866]:

$$a_k = -a_k^n \big|_{n=M} \tag{7}$$

where

$$a_k^n = a_k^{n-1} - \rho_k a_{n-k}^{n-1}, \quad k = 0, \cdots, n \text{ and } n = 0, \cdots, M \tag{8}$$

and $a_0^n = -1$ and $a_q^p = 0$ if $p < q$.

IV. Discussion

It is of special interest to point out the incremental nature of the PARCOR algorithm. This nature gives to it a great power for the numerical solution of propagation equations in nonuniform, nonhomogeneous media. This analytical procedure may be of great interest in the measurement and characterization of the acoustical transmission behavior of non-easily accessible structures like the ear. It seems to be possible to obtain measurements about the transmission characteristics of the ear by means of simple and inexpensive systems based on the deconvolution of the reflections collected in the outer canal of the ear, after the stimulation of the auditory system, while it may also be possible to obtain information about the characteristics of the bony chain and even of cochlear properties in this way. The proposed hearing aids could use the model in two different ways, one being to correct for mechanical and conductive impairments and the other to correct for the total auditory system including the neural pathways. The first one can be implemented using the filter coefficients adapted from objective measurements in the model as discussed and calculated here, while in the second approach subjective criteria of hearing improvements felt by the hearing-impaired person should be used, this probably leading to a more subjectively useful hearing aid.

As can be appreciated from the complicated nature of the ear, it is anticipated that nonlinear effects may play an important role. As far as the mechanical behavior of the auditory system is concerned, although several field researchers have reported almost linear or quasi-linear measurements [21], [22] for cochlear mechanics, the problem looks strongly nonlinear for the mechanical-to-neural transduction and for neural transmission to the brain. Thus it is felt that the linear theory treated here could be improved upon by application of recent nonlinear treatments [23]. For sure, it was due to a discussion with Professor Parker, which we wish to acknowledge here, that we were led to the above considerations.

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Book Reviews

The following reviews were selected from those recently published in various IEEE TRANSACTIONS *and Group/Society Magazines and Newsletters. They are reprinted here to make them conveniently available to the many readers who otherwise might not have ready access to them. Each review is followed by an identification of its original source.*

Quality Assurance for Computer Software–Robert Dunn and Richard Ullmann (New York: McGraw-Hill, 1982, 351 pp., $24.50). *Reviewed by Chris N. Napjus, Department of Defense, Ft. Meade, MD.*

This lucid, well-organized, and immensely readable book covers an astounding breadth of material in its 351 pages. A great many of the topics discussed, viewed separately, might not ordinarily be identified with quality assurance functions. Yet the authors' low-key message, communicated through the logical progression of topics, is that all aspects of software are–and must be–considerations in a proper quality assurance program.

Addressing a broad range of topics in a limited space involves an unfortunate corollary: a restriction on the amount of detail that can be presented with each subject. It is very difficult to determine the book's intended focus or the most appropriate audience. The majority of the book is written at a tutorial level, yet the last chapter goes into excruciating mathematical detail. The early chapters are intended for software neophytes or people who possess a hardware quality assurance background, yet the end is surely of interest to only a small percentage of professional software QA personnel.

The bulk of the book can be appreciated by most software practitioners, but even here there is a problem. There is too little detail on most topics to consider this a "how-to" exposition on software quality assurance. In order to fully appreciate the importance of many subjects that are discussed very briefly or referenced only by name, one should possess a broad software engineering background. As the authors note, this is seldom the case with software personnel.

Despite these perceived problems of scope and focus, there is little to criticize about the writing. The authors' command and use of English offers a refreshing contrast to much of the current writing on technical subjects, and the text is interesting–often engrossing. There are enough informational nuggets throughout to attract readers of vastly different backgrounds, although those with less experience may miss much of what is intended while more knowledgeable readers will see familiar material packaged in a fresh manner.

The book is divided into five parts. The first, consisting of a short introductory chapter, clearly establishes the need for software quality assurance. It points out the key differences between software and hardware QA needs in as concise and understandable manner as one could hope for.

Part 2 provides three chapters on the nature of computer software, which the authors recommend "to readers who have little experience with software." I am not sure the book needs these chapters, but I still recommend them for their clear exposition of an inherently difficult subject. Chapter 3 in particular, on the life cycle of software, is superb in its use of a progressive example. Even if readers don't require this technical refresher, they will learn something about writing style.

The real meat of the book appears in the five chapters that comprise Part 3. Successive chapters on defect prevention, configuration management, testing, and tools provide individually self-contained treatments of these important topics while simultaneously utilizing the preceding material for reinforcement. These four chapters provide the necessary background for Chapter 9, "The Quality Program." This chapter alone covers the bulk of the topics one might have expected to find in the entire book based on its title. Included in this chapter are discussions on the role of quality, its potential benefits, its concepts and basic techniques, and relevant educational aspects.

Part 4, on the implementations of a QA program, might have included Chapter 9 as well. Chapter 10 discusses the problem of selling a QA program to management, along with strategies for assembling and training the required staff. Standards and policies, both existing and required, and the proper content of a QA plan are examined in Chapter 11.

Finally, Part 5 contains a single chapter on reliability and the measurement of goodness. This chapter is out of balance internally because it offers little on the desirability of quantitative measures and a great deal on various reliability models; it is also out of balance with the remainder of the book because it reverses its approach by treating a narrow sub-