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SIM LE DIGITAL SPEECH SYNTHESIS

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SUMMARY

Relatively simple computer methods for synthesizing speech to be used in phonetic/perceptual research are presented, with particular references to the problems and successes encountered in the development of such a system at Central Institute for the Deaf and the Biomedical Computer Laboratory of Washington University. The purpose of this paper is to present a synthesis and clarification of established methods so as to encourage other computational linguists to tackle digital speech synthesis. The approach is semi-tutorial: crucial algorithms are given in Fortran or block-diagram form, and bibliographic references that were found to be most useful in the system development are listed and discussed.

The system described requires a minimum of hardware; a minicomputer is sufficient, if it is equipped with tape or disk secondary memory. The sound pressure wave is calculated entirely by software and only a digital-to-analog converter and a low-pass filter are required to convert it to a recordable electrical signal.

The vocal apparatus is simulated by a rough model which is still general enough to make most speech sounds. The two main types of excitation of the vocal tract -- periodic glottal waves for voicing and random noise for frication or aspiration -- are supplied by algorithms presented as function subroutines. The effect of the vocal tract on these inputs is modelled by combinations of three other elemental functions, whose coding is based on recursive equation theory for computational efficiency. A resonance provides the user with a means for accentuating the signal at a certain frequency, such as a formant frequency of a vowel; an anti-resonance or notch filter is provided to cut back the energy at a certain frequency, as in simulation of the nasal anti-formant; and a radiation-effect subroutine simulates the effect on the speech signal of passage from the lips through a short stretch of air. Empirically obtained wave shapes and spectra of the outputs of these five basic functions are given in order to give the reader a better feel for what they do.

These five elements can be combined in a number of ways. A detailed discussion is given for one of the simplest reasonable models, in which the glottal wave and frication generators excite a series of three variable resonators, using a set of fixed resonances to simulate higher frequency formants in addition to the radiation-effect simulator. Several other more complex arrangements are presented, including parallel resonator models and models with separate filters for shaping voiced, fricative, and nasal components, and their advantages and disadvantages are discussed.

An example is given of a complete modular Fortran program generating the word "seat". The equations for specifying the parameters controlling the elemental functions were derived, with much effort, from analysis of one token utterance, and spectrographs of the real and synthetic words are shown to illustrate the degree of naturalness obtainable with the simple three-resonator series model. A simpler example for generating a constant vowel sound is also given, along with a summary of data useful in making many vowels.

This paper is a slightly expanded version of one given orally at the 12th annual A.C.L. meeting in Amherst Massachusetts.

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I. Introduction

Several years ago, it was decided that the Research Department at Central Institute for the Deaf in St. Louis should have a digital speech synthesizer to aid studies in psychoacoustics and phonetic perception. The equipment on hand at the time consisted primarily of a 12-bit-word mini-computer with keyboard and scope, with special-purpose hardware for doing digital-to-analog conversion, low-pass filtering, and floating-point arithmetic More peripheral devices and core memory have now been added. We have been working since then on writing digital computer programs to synthesize speech studying the literature and gaining practical experience.

Almost all of the theory and techniques necessary to program the synthesis of English sounds can be found in published literature, but in bits and pieces, here and there. Utilizing the contributions of many authors, plus our own experience, we present and explain a basic program for synthesizing speech, in the hope that computational linguists who have not worked with low-level speech phenomena may be encouraged to program synthesizers.

The use of synthetic-speech stimuli has been extremely important to the investigation of the perceptually distinctive features of speech and of low-level phonological rules, but much work remains undone. Synthesizing speech is clearly important to phonetic research, and the field could well use more researchers with linguistic training. The system described in this paper requires relatively small investments in equipment and programming.

II. Overview

Synthesizers use varying amounts of special-purpose hardware. The type of synthesis we describe here uses the bare minimum, calculating the speech wave on a digital computer and requiring only a digital-to-analog converter and a low-pass analog filter as special hardware. This minimum set of equipment is shown in Figure 1, page 6. If we assume that tape or disk secondary storage is available, then a 4 k 12-bit-word mini-computer is sufficiently large, and a 12-bit D/A converter will give enough dynamic range.

Once the speech wave is generated and stored on tape or disk, there is a problem of writing a program to output enough of it synchronously at a fast enough rate. We will not go further into this problem here, since the solution will depend on the particular machine installation you have. Whatever output sample rate you achieve, there are two things to note: the analog low-pass filter should pass only frequencies below the output sample rate, and the output sample rate is a parameter whose value must be fed into the digital calculations.

Although our synthesizer can be described as a terminal analog model of the vocal apparatus, the attitude we take is that our method of synthesis is used rather to produce the significant acoustic features of speech. To set the stage for an understanding of the synthesizer presented here, and for the benefit of those not familiar with acoustic phonetics, let us review for a minute what we are synthesizing.

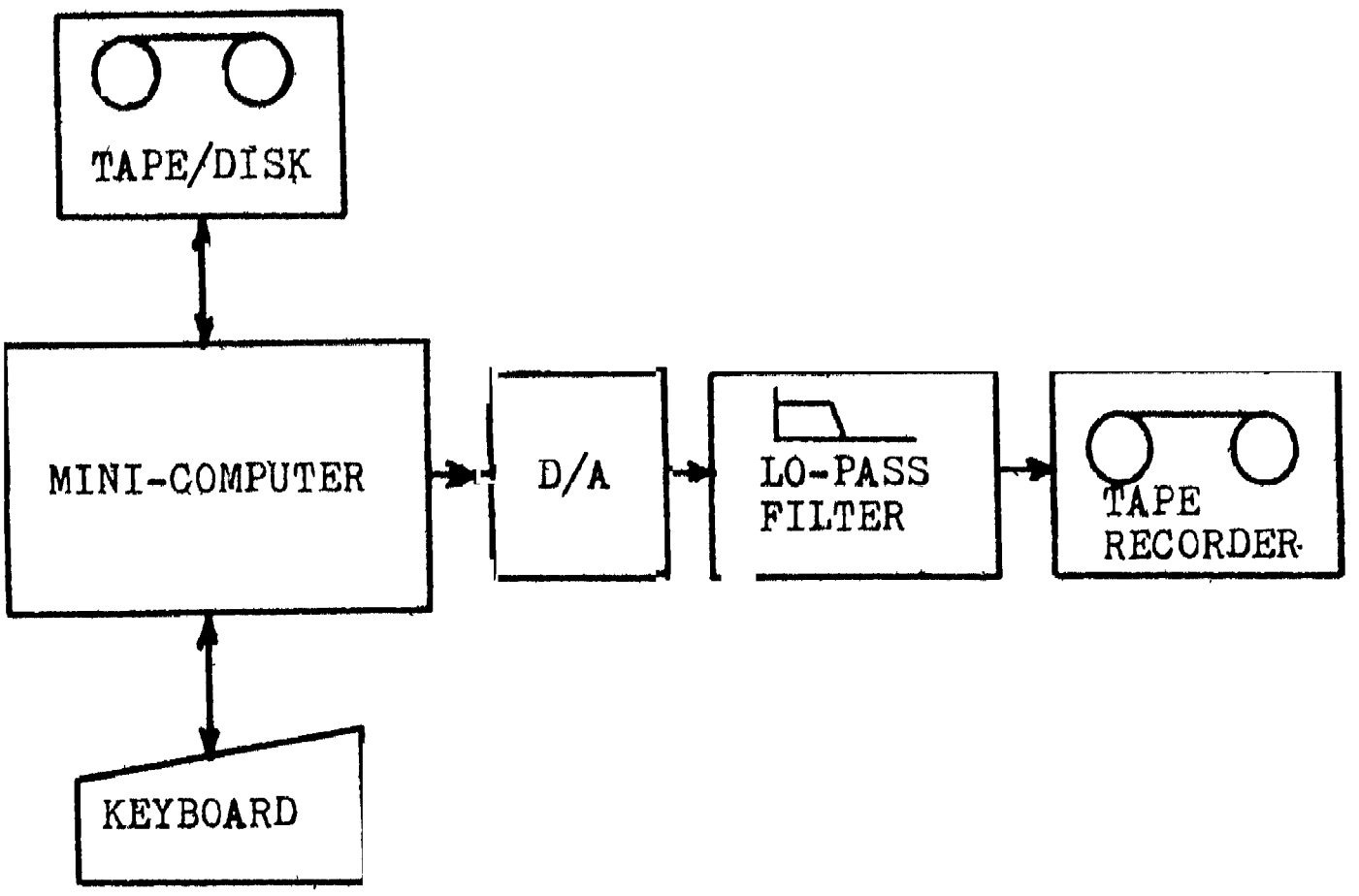

Figure 1. Minimum Hardware Needed

Figure 2, page 8, shows a typical spectrogram of the word "seat." Frequency is the vertical axis, time the horizontal, and darker marks indicate more energy. The mottled area at the upper left is the quasi-random noise of the /s/ sound. To make other fricatives, such as /f/, we need to alter the frequency spectrum and intensity of the noise. The dark horizontal bands in the center are concentrations of energy -- resonances called "formants" -- characteristic of vowel sounds. To make different vowels, we need to change the center frequencies, bandwidths, and relative intensities of the three lower formants visible here. There are higher-frequency formants, which do not show up well in this spectrogram, but they seem to be important only to the naturalness of the speech, not to which vowel is perceived. Note the beginning and ending slopes of the formants; these formant transitions are crucial to the perception of occlusive consonants such as the stops /b/, /d/, and /g/. And finally, the vertical mark at the far right is a burst of noise marking the release of the final consonant.

The detailed algorithms we present here will be expressed in Fortran for clarity, although the synthesizer with which we have had the most experience is coded in assembly language. We are presently converting to Fortran, and the subroutines and final example program listed in this paper have been tested in their Fortran form.

Before we get into details of synthesis, consider the overall logic, Figure 3, page 9. The main thing to note in

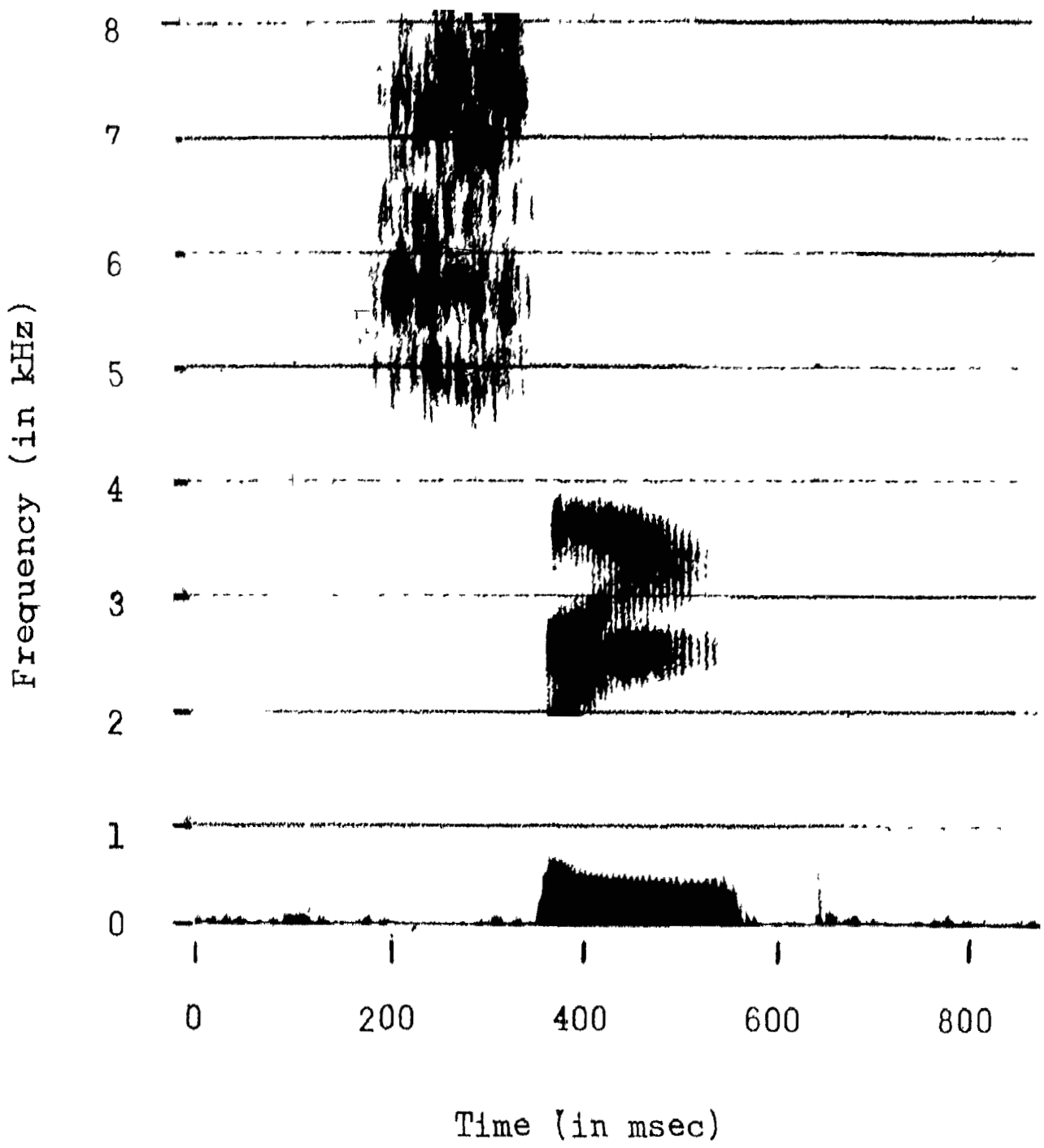

Figure 2. Typical Spectrogram of "Seat"

```
C  SKELETON OF SYNTHESIS PROGRAM
C
      DIMENSION NBUF(256)
C
C  INITIALIZE
C
      NBSIZE=256
      NBLKS=10
C
C  GENERATE
C
      DO 800 NB=1,NBLKS
      DO 750 NPT=1,NBSIZE
C
C  GENERATE NEXT SPEECH WAVE POINT
C  AS THE VALUE OF YN
C
C
C  STORE SPEECH WAVE POINT
C
      NBUF(NPT)=IFIX(YN)
750   CONTINUE
C
C  WRITE OUTPUT BUFFER
C
      CALL WRITB(NBUF)
800   CONTINUE
      CALL EXIT
      END
```

Figure 3. Over-all Synthesizer Program Logic.

WRITB is a subroutine to write the contents of the array NBUF onto tape or disk.

this logic is that we synthesize the speech a point at a time, in one pass, storing each buffer-full on tape or disk as it is generated. This program will produce an integral number of buffer-loads, but other methods for terminating the main loop are easy to implement.

III. Basic Elements

There are five basic elements in this method of speech synthesis:

1. A glottal wave generator, with controls for repetition rate (pitch) and amplitude;
2. A white-noise generator, with a control for amplitude;
3. A resonant filter, with center frequency and bandwidth controls;
4. An anti-resonant filter, with similar controls;

and 5. A radiation-effect simulator.

These five elements can be connected in a variety of ways to produce models of greater or lesser complexity. The glottal wave generator and noise generator produce sounds whose spectra are then shaped by combinations of the other elements.

A. Sources

1. Glottal-Wave Generator

Natural glottal waves, while subject to much variation, are usually considered to consist of three parts: a glottis-opening phase in which the volume velocity is increasing, a glottis-closing phase in which the volume velocity is decreasing, and a glottis-closed phase in which the volume velocity is zero. The spectrum of such waves is supposed to fall off at about 10 to 12 dB/octave,[1]

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1 Cf. Flanagan (1958)

and a theoretical spectral analysis of a linear approximation shows regular lobes superimposed on this spectrum.[1] We have just begun to study natural glottal waves in our Research Department, and while no definite results can be reported yet, Figure 4, page 12, shows the wave shape and spectral analysis of one typical male glottal wave period. The opening and closing phases will be more apparent if the wave is considered to begin and end at the minimum value instead of a zero value as our analysis program has done. There does not appear to be a closed phase of the wave, but this is one of the variations reported in the literature.

Two general methods have been used to produce approximations to glottal waves: in the first, an impulse is generated and fed into appropriate spectral shaping filters,[2] while in the second, a standard wave shape is used as a pattern for each glottal pulse.[3] We chose the second method, because Rosenberg has shown that some simple wave shapes produce synthetic speech that sounds natural,[4] and the method seemed conceptually cleaner.

Rosenberg studied the naturalness of speech synthesized from a number of different wave shapes used as glottal pulses. We have used two of his shapes in our work: the linear (called by some "triangular") and the polynomial approximations. Of the shapes he studied, the linear requires the least amount of

---

1. cf. Flanagan (1958) and Dunn et al. (1962)
2. cf. Rabiner (1968) and Fant and Martony (1962)
3. cf. Rosenberg (1971) and Sekimoto (1973)
4. Rosenberg (1971) reports that some subjects preferred synthetic speech with certain glottal wave shapes to natural speech.

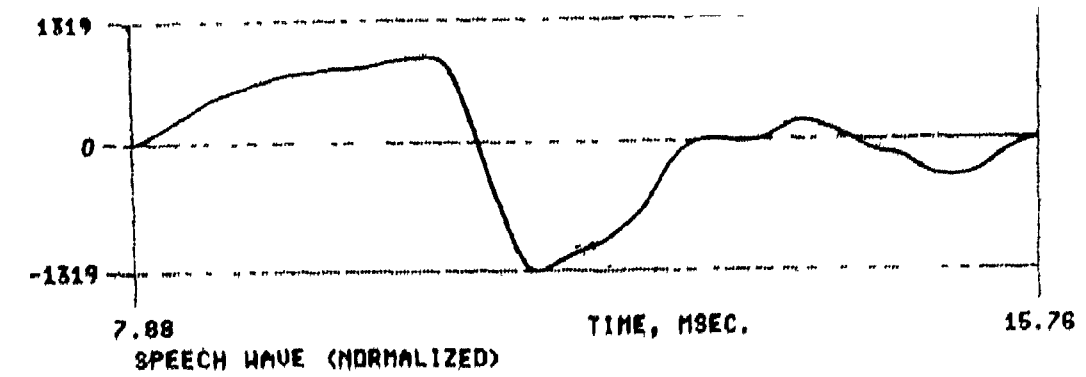


Figure 4. Wave Shape and Spectral Analysis of a Typical Glottal Wave Period. The intensity spectrum has been normalized so that the largest component is at the 80 dB level; before normalization it was 61.54 dB relative to an arbitrary standard of 0.25. The horizontal line across the intensity spectrum graph is a conservatively estimated noise cut-off line; any component with magnitude less than this level may be the product of computational noise. The phase angles of the components are $\emptyset_n$, where the wave is represented by $\sum_n A_n \sin(nw_1t + \emptyset_n)$ and $\emptyset_1$ is arbitrarily zero. The phase angle display is suppressed for any component whose intensity falls below the noise cut-off line.

execution time and the polynomial scored the highest in his tests of naturalness. Figure 5, page 14, shows the wave shape and spectrum of a linear approximation and Figure 6, page 14, shows that of a polynomial approximation. The over-all falloff of the spectrum of the polynomial more nearly matches our example wave, but no lobes are apparent, as they are in the spectra of the linear approximation and the natural example. Judging from some informal listening tests we have made, these distinctions do not seem to make a great difference: both approximations sound good.

As Figure 7, page 15, we present a Fortran function GLOT(P,AV) for generating glottal waves using the polynomial approximation. Values of pitch (P) and zero-to-peak amplitude (AV) are passed directly as control parameters.

The subroutine uses the common area to store several variables, which or course could be declared as formal parameters instead. ISWV is a voicing switch used in the logic internal to GLOT. TDEL is the period between output sample points in milliseconds; in the over-all initialization of the program, its value should be calculated as 1000.0/OSR, where OSR is the output sample rate in samples per second. TG is a variable used as a simulated time clock by GLOT, keeping track of how far through the glottal wave it has gone. TP, T1, and T2 are durations (in milliseconds) from the beginning or the glottal pulse, calculated and used by GLOT: TP is the duration of the pulse, T1 is the duration of the opening phase, and T2 is the duration of the opening and closing phases combined. OPTR (an acronym for "opening time

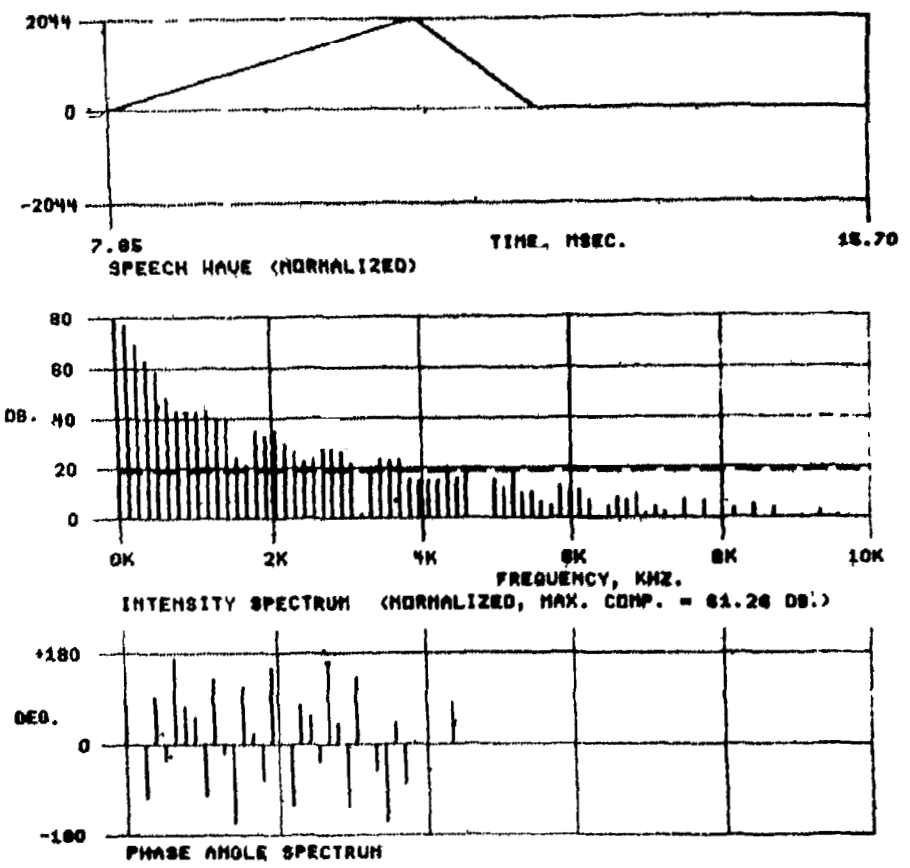


Figure 5. Wave Shape and Spectral Analysis of a Linear Approximation to a Glottal Wave. For details of the display, cf. Figure 4.

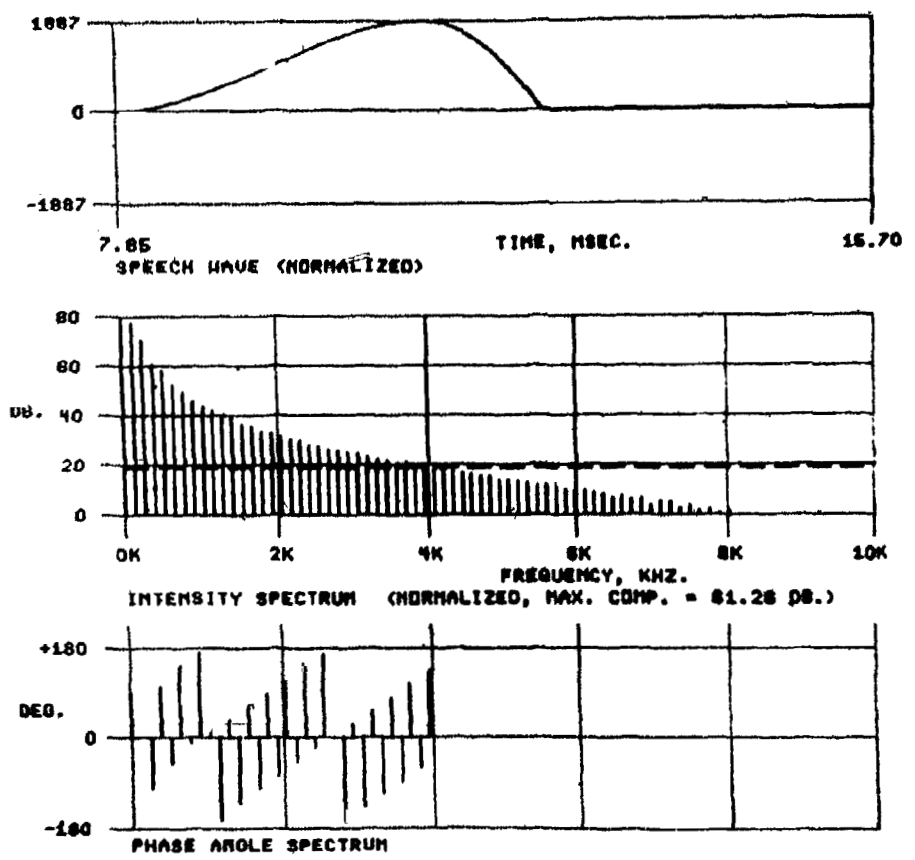


Figure 6. Wave Shape and Spectral Analysis of a Polynomial Approximation to a Glottal Wave. For details of the display, cf. Figure 4.

```
0001            FUNCTION GLOT(P,AV)
0002            COMMON ISWV,TDEL,TG,TP,T1,T2,OPTR,CLTR,AVSAVE
0003      C
0004      C PRODUCES A POLYNOMIAL APPROXIMATION TO A GLOTTAL WAVE
0005      C WITH CONSTANT WAVE SHAPE
0006      C
0007      C USES THE FOLLOWING VARIABLES FROM COMMON: ISWV,
0008      C TDEL,TG,TP,T1,T2,OPTR,CLTR,AVSAVE
0009      C
0010      C FIRST SEE IF WE NEED TO RE-INITIALIZE.
0011      C FOR BEGINNING OF GLOTTAL WAVE
0012      C
0013      C ARE WE NOW GENERATING VOICE?
0014            IF (ISWV) 900,20,10
0015      C YES -- IF WE ARE NOT AT THE END OF A PULSE,
0016      C PARAMETERS ARE O.K.
0017      C OTHERWISE WE NEED TO CHECK AV TO SEE
0018      C IF WE NEED ANOTHER PULSE
0019      10    IF (TG+0.5*TDEL-TP) 50,20,20
0020      C EITHER WE HAVE NOT BEEN GENERATING VOICE OR
0021      C WE HAVE JUST FINISHED A PULSE --
0022      C IF AV > 0, INITIALIZE TO GENERATE A(NOTHER) PULSE
0023      C AND RESET ISWV TO 1
0024      C OTHERWISE (RE)SET ISWV TO 0
0025      20    IF (AV) 30,30,40
0026      30    ISWV=0
0027            GO TO 50
0028      C INITIALIZE FOR ANOTHER PULSE
0029      40    ISWV=1
0030            AVSAVE=AV
0031            TG=0.0
0032            TP=1000.0/P
0033            T1=OPTR*TP
0034            T2=T1+CLTR*TP
0035      50    CONTINUE
0036      C
0037      C END OF PARAMETER SETTING
0038      C
0039      C BEGINNING OF LOGIC TO GENERATE GLOTTAL WAVE
0040      C
0041            IF (ISWV) 57,55,57
0042      55    Y=0.0
0043            GO TO 180
0044      57    CONTINUE
0045      C
0046      C IF TG < T1, Y=AV*(3*(TG/T1)**2-2*(TG/T1)**3)
0047      130   IF (TG-T1) 140,150,150
0048      140   Y=AVSAVE*(3.0*((TG/T1)**2)-2.0*((TG/T1)**3))
0049            GO TO 180
0050      C ELSE IF TG < T2, Y=AV*(1-(((TG-T1)/(T2-T1))**2))
0051      150   IF (TG-T2) 160,170,170
0052      160   Y=AVSAVE*(1.0-(((TG-T1)/(T2-T1))**2))
0053            GO TO 180
0054      C ELSE Y=0
0055      170   Y=0.0
0056            GO TO 180
0057      C
0058      C END OF GLOTTAL PULSE GENERATION
0059      C RETURN VALUE OF GLOTTAL WAVE
0060      C AND INCREMENT TG
0061      C
0062      180   GLOT=Y
0063            TG=TG+IDEL
0064            RETURN
0065      C
0066      C ERROR IN VALUE OF ISWV
0067      C
0068      900   WRITE(1,910)ISWV
0069      910   FORMAT(" ** ERR IN GLOT1: ISWV=",I5)
0070            CALL HOLD
0071            CALL EXIT
0072            END
0073      C GLOT
```

Figure 7. A Fortran Function to Generate Glottal Waves

ratio") is the fraction of the wave occupied by the opening phase, and CLTR is the fraction occupied by closing. In the over-all initialization, OPTR should be set to .40 and CLTR to .16, values which maximize naturalness according to Rosenberg's paper.

If the instantaneous values of AV were used by GLOT, the standard wave shape would be altered if AV were changing during generation of a glottal wave. To keep the wave shape constant, GLOT uses the variable AVSAVE to store the value of AV at the beginning of each pitch period, and during the generation of the pulse, AVSAVE is used as the (constant) amplitude.

Between calls to GLOT, the values of ISWV, TG, TP, T1, T2, and AVSAVE should not be altered.

Rosenberg's equations for the polynomial approximation are used in GLOT; to get the linear approximation, the following two lines of Fortran should be substituted in GLOT for lines number 60 and 64:

```
140     Y=AVSAVE*(TG/T1)
160     Y=AVSAVE*(1.0-(TG-T1)/(T2-T1))
```

2. White Noise Generator

Almost any reasonably good random-number generator can be used as a source of white noise. If the spectrum of the random numbers produced is flat, it will be easier to shape into the desired spectra for the different fricative sounds.

The algorithm we use was developed for use in synthetic speech work; it is very fast, and produces noise with a quite flat spectrum. Its presentation by Rader, Rabiner, Schafer,

and Perry[1] is easy to follow and implement. The logic of the algorithm is formally stated in Fortran in Figure 8, page 18, but if possible this should be one function coded in assembly language: if the right machine instructions are available, it will be a snap, but "bit fiddling" in Fortran is very slow.

Our function IRN4(X) -- X is a dummy variable required by our Fortran compiler -- contains this algorithm in assembly language, producing on successive calls a series of random numbers with a uniform distribution over the interval from -2047 to +2047. To implement a white noise generator, only this line of coding is needed:

```
Y=AN*(FLOAT(IRN4(X))/2047.0)
```

where AN is a variable whose value is the amplitude of noise desired.

A typical stretch of noise produced in this manner, along with its spectrum, is given as Figure 9, page 19. Note that there does not appear to be any significant deviation from flatness in the spectrum intensity.

B. Spectral Shaping Elements

1. Resonances and Anti-resonances

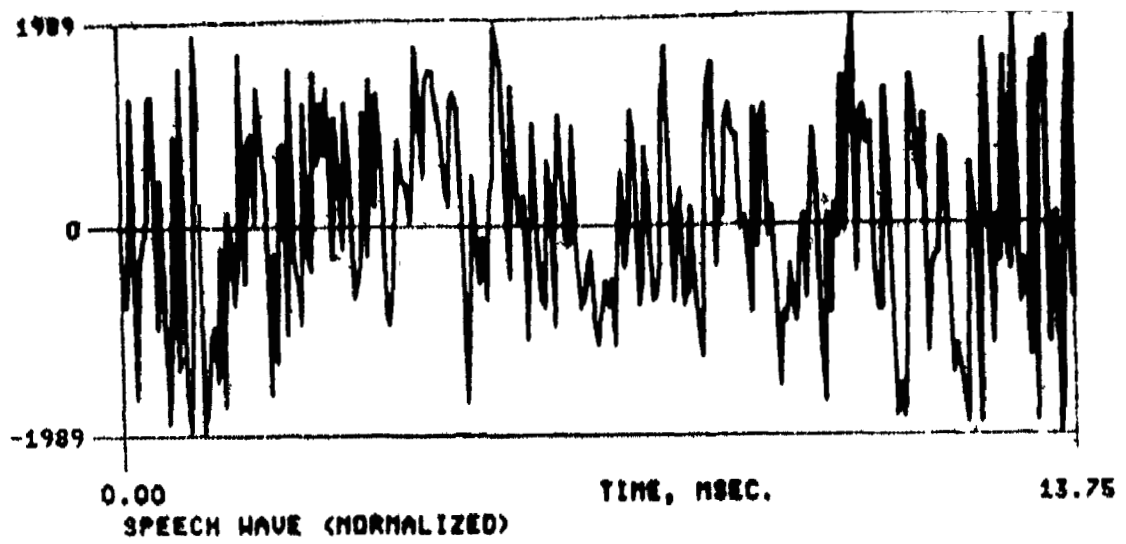
We use recursive equations, a technique developed by electrical engineers, to simulate resonant and anti-resonant (notch) filters as elements to shape spectra. Each individual filter can be represented by a second-order linear differential

---

1. Rader, Rabiner and Schafer (1970) and Perry, Schafer, and Rabiner (1972)

```
      FUNCTION IRN4(X)
      COMMON NM1(19), NM2(19)
      DIMENSION NX1(19),NX2(19)
C FORM BIT-WISE EXCLUSIVE OR OF NM1,NM2
      DO 10 I=1,19
10    NX1(I)=MOR(NM1(I),NM2(I))
C ROTATE NX1 8 PLACES TO THE RIGHT
      DO 20 I=1,11
20    NX2(I+8)=NX1(I)
      DO 30 I=12,19
30    NX2(I-11)=NX1(I)
C SHIFT PAST VALUES
      DO 40 I=1,19
      NM2(I)=NM1(I)
40    NM1(I)=NX2(I)
C RETURN VALUE OF LEFT-MOST 12 BITS OF NX2
      IRN4=MINT(NX2)
C END
      RETURN
      END
```

Figure 8. Random Number Generator Documented in Fortran. NM1(19) and NM2(19) are arrays whose elements have either the value 0 or 1. MOR(N1,N2) is a function returning the exclusive or of N1 and N2, variables having either the value 0 or 1. MINT(N) is a function whose argument N is a bit-string array such as NM1 and which returns a 12-bit integer value consisting of the left-most 12 elements of N packed into a single word.



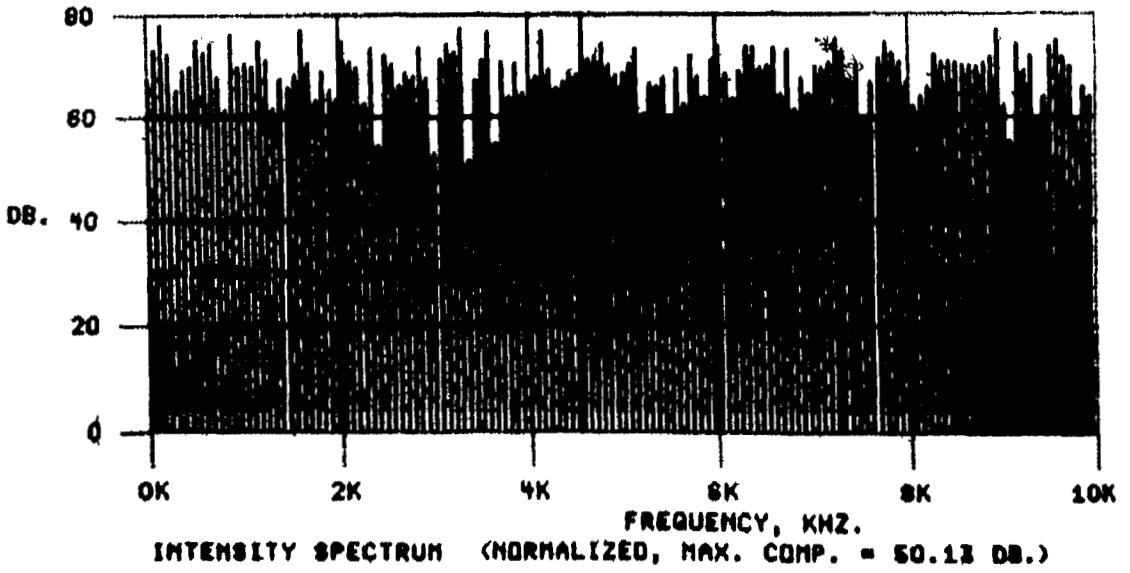


Figure 9. Typical Wave Form and Spectral Analysis of the Output From the White Noise Generator. For details of the display cf. Figure 4.

equation, giving only one resonance or anti-resonance. For those who feel at home in the s-plane, the recent book Speech Synthesis edited by Flanagan and Rabiner contains reprints of papers developing the theory of recursive equation filter simulation: for the rest of us, the paper by Lovell et al. (1973) is a clear presentation, with some more general Fortran algorithms than will be given here.

Figure 10, page 21, gives one Fortran subroutine and two Fortran functions which we use to simulate resonant and anti-resonant filters.

The functions RES and ARES return the output values of simple resonant (conjugate pole pair) and anti-resonant (con jugate zero pair) filters, respectively. A0, A1, and A2 are coefficients used in the recursive equations. YM1 and YM2 are remembered previous values of the signal, and Y is the input to the filter. A0, A1, and A2 determine the characteristics of the filter: center frequency and bandwidth. Each simulated filter should have its own variables in which to save the values of YM1 and YM2, and between calls to the function simulating that filter, the values of these variables should not be changed.

The subroutine COEFF is used to calculate appropriate values for A0, A1, and A2, based on CF, the center frequency, and BW, the bandwidth, of the resonance or anti-resonance.[1] SR is the output sample rate, and MPZ tells the subroutine

---

1. In the version of this paper presented at the 12th annual A.C.L. meeting, there was an error in line 16 of subroutine COEFF

```
0001                SUBROUTINE COEFF(CF,BW,A0,A1,A2,SR,MPZ)
0002          C
0003          C         COMPUTES THE RECURSIVE EQUATION COEFFICIENTS
0004          C   A0,A1,A2 FOR EITHER A RESONANT OR ANTIRESONATE FILTER
0005          C   SPECIFIED BY CENTER FREQUENCY CF AND BANDWIDTH BW (HZ.)
0006          C   SR IS THE SAMPLE RATE (SAMPLES/SEC)
0007          C   IF MPZ=1, FILTER WILL BE A RESONANCE
0010          C   IF MPZ=0, FILTER WILL BE AN ANTIRESONANCE
0011                PI=3.14159265
0012                A=PI*BW/SR
0013                B=2.0*PI*CF/SR
0014                A2=EXP(-2.0*A)
0015                A1=2.0*EXP(-A)*COS(B)
0016                A0=1.0-A1+A2
0017                IF (MPZ) 20,10,20
0020          10    A0=1.0/A0
0021          20    RETURN
0022                END
```

```
0001                FUNCTION RES(Y,YM1,YM2,A0,A1,A2)
0002          C
0003          C         SIMULATES RESONATOR (CONJUGATE POLE PAIR)
0004          C   GIVEN,BY RECURSIVE EQUATION COEFFICIENTS A0,A1,A2
0005          C   YM1 AND YM2 ARE PAST VALUES OF Y; THEIR VALUES
0006          C   MUST BE SAVED
0007                YRES=A0*Y+A1*YM1-A2*YM2
0010                YM2=YM1
0011                YM1=YRES
0012                RES=YRES
0013                RETURN
0014                END
```

```
0001                FUNCTION ARES(Y,YM1,YM2,A0,A1,A2)
0002          C
0003          C         SIMULATES ANTIRESONATOR (CONJUGATE ZERO PAIR)
0004          C   GIVEN BY RECURSIVE EQUATION COEFFICIENTS A0,A1,A2
0005          C   YM1 AND YM2 ARE PAST VALUES OF Y; THEIR VALUES
0006          C   MUST BE SAVED
0007                TEMP=Y*A0
0010                ARES=TEMP-A1*YM1+A2*YM2
0011                YM2=YM1
0012                YM1=TEMP
0013                RETURN
0014                END
```

Figure 10. Fortran Implementation of Elemental Filters

whether to compute coefficients for a resonance or an anti-resonance

The spectral effect of a resonance with center frequency of 3000 Hz and bandwidth of 200 Hz is illustrated in Figure 11, page 23, while Figure 12, page 23, is a similar illustration of the effect of an anti-resonance of the same center frequency and bandwidth. In these figures, the first graph shows output for an impulse input and the second graph shows the normalized intensity spectrum of that output.

2. Radiation Effect

The effect on the spectrum of the radiation of sound from the lips through a short stretch of air can be reasonably approximated by a differentiator.[1] Figure 13, page 24, gives a simple Fortran function RAD simulating this effect. Y is the speech wave input to the simulater, YM1 is the remembered immediately previous value of Y, and G is a normalizing gain control which should be calculated in the over-all initialization as a direct function of the output sample rate, some K times OSR. The spectral effect of RAD, approximately a 6 dB/octave rise, is illustrated in Figure 14, page 24.

IV. Organization of Elements

A. The Simplest Model

The simplest reasonable model for connecting these elements, which we have taken to calling "Model T", is given in block diagram form in Figure 15, page 25. We use this organization in our currently running synthesizer. The three variable

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1. Rabiner (1968) p. 823

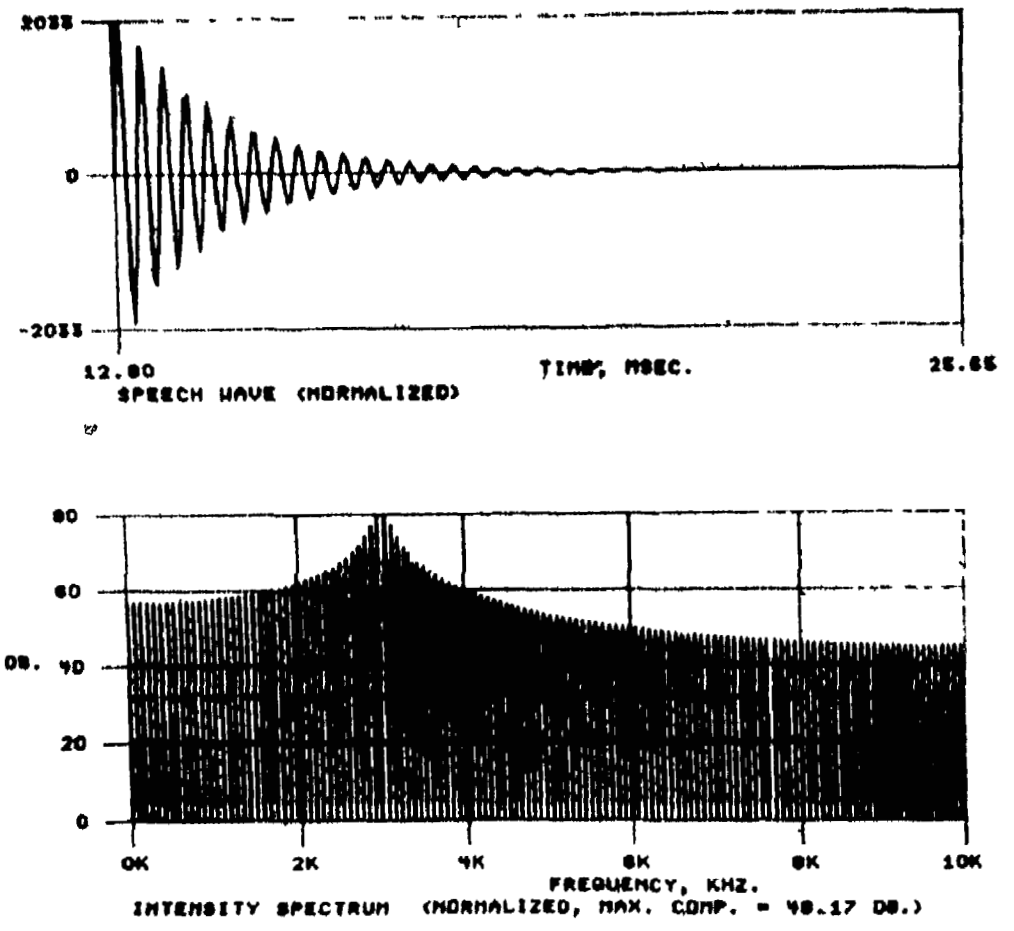

Figure 11. Spectral Shaping Effect of an Elemental Resonant Filter with CF=3000 Hz and BW=200 Hz.

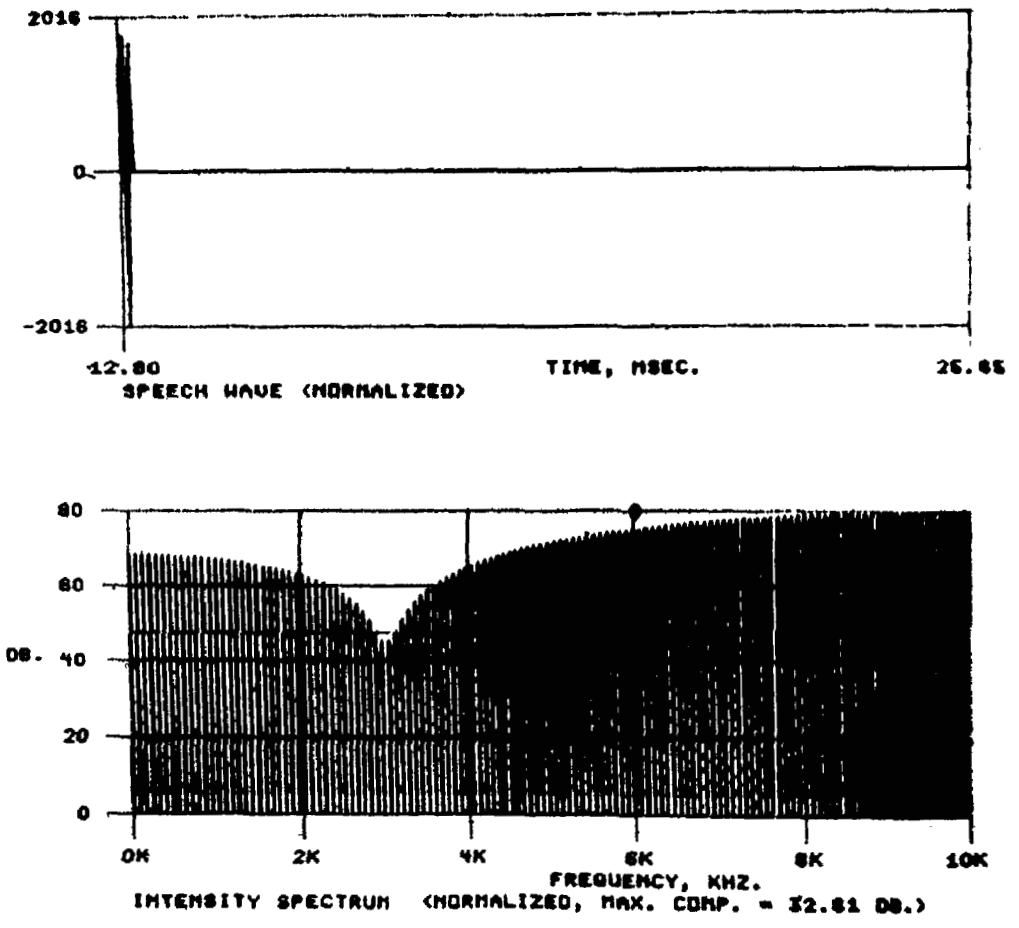

Figure 12. Spectral Shaping Effect of an Elemental Anti-Resonant Filter with CF=3000 Hz and BW=200 Hz.

```
0001                FUNCTION RAD(Y,YM1,G)
0002          C
0003          C     SIMULATES RADIATION EFFECT
0004          C     BY TAKING FIRST TIME DERIVATIVE OF Y
0005          C
0006                RAD=G*(Y-YM1)
0007                YM1=Y
0010                RETURN
0011                END
```

Figure 13. Fortran Function to Simulate Radiation Effect.

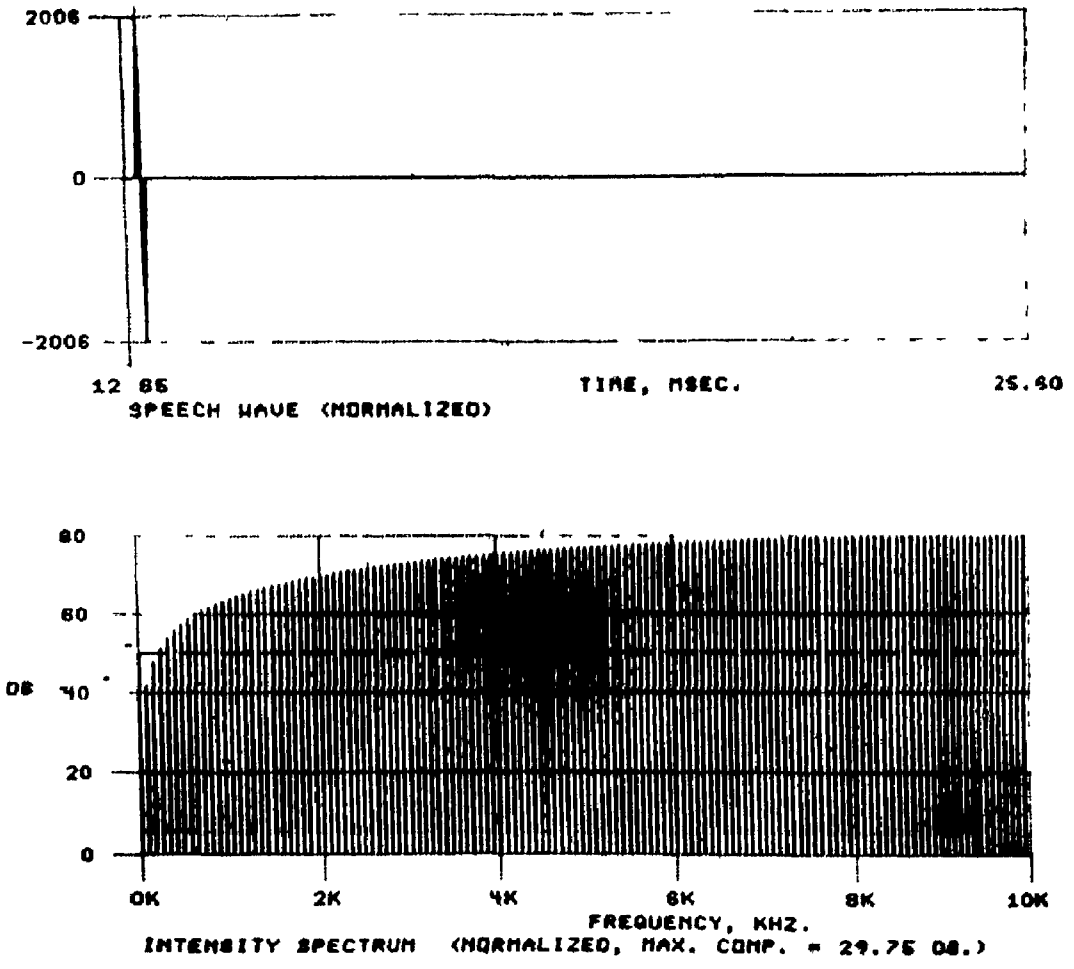

Figure 14. Spectral Shaping Effect of Radiation as Simulated by Function RAD.

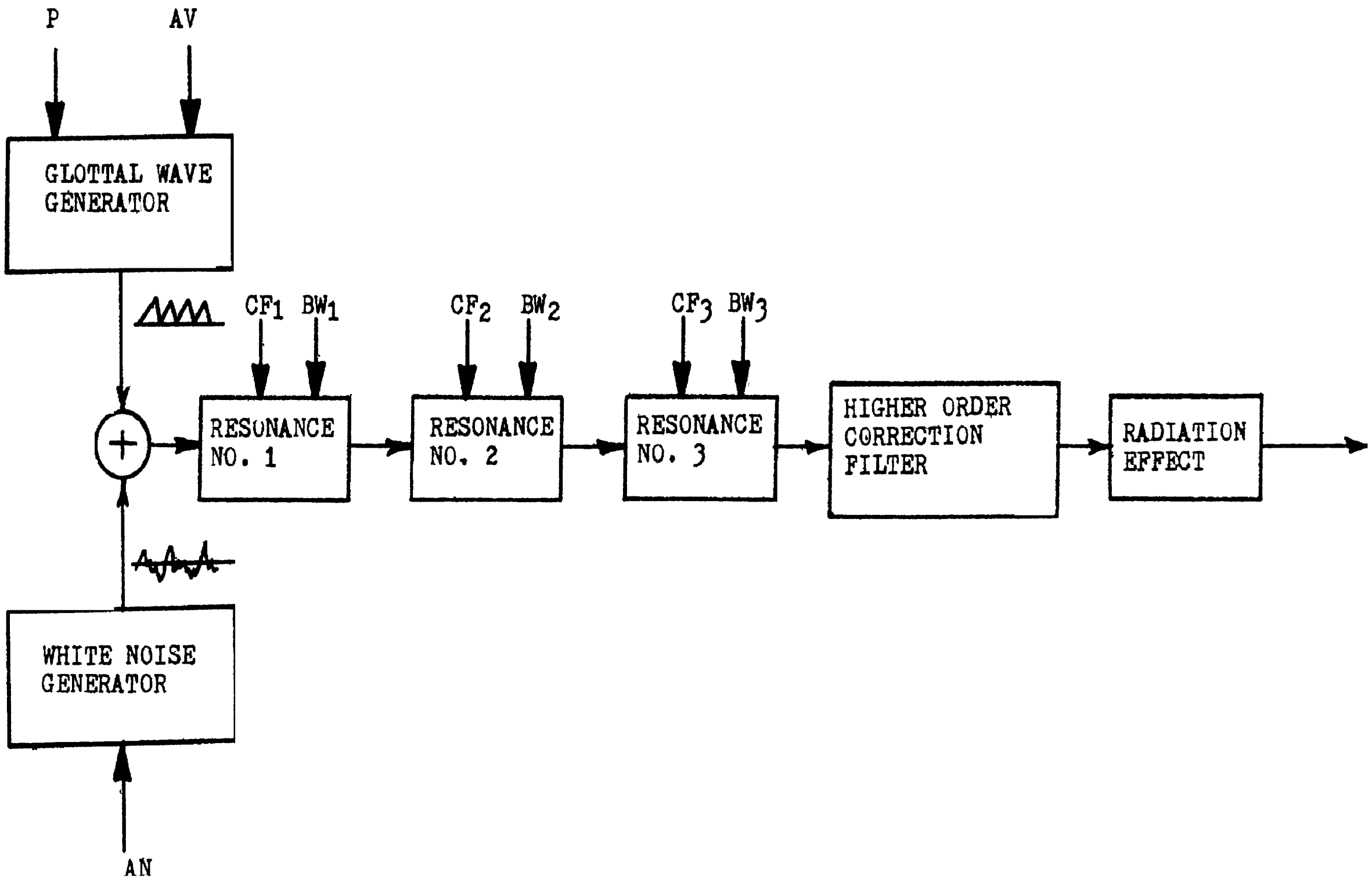

Figure 15. "Model T" Organization Block Diagram

resonant filters are used to make the formants of voiced speech and, in a rather strained fashion, to shape the noise spectrum during frication and aspiration.

All of the elements in this figure should now be familiar except the one called "higher order correction filter." This is a series of resonant filters of fixed center frequency and bandwidth which compensate for the effect of higher-frequency resonances present in a real vocal tract but absent in a digital simulation of this kind. Their use is discussed in Rabiner (1968) from which the values presented in Figure 16, page 27, were taken. These are the values to use for center frequency and bandwidth of the higher order correcting filters. Only higher-order filters with center frequency less than ½ the output sample rate should be used. The recursive equation coefficients A0, A1, and A2 need be calculated only once, in the over-all initialization.

Theoretically, the order of computation of the series elements such as those in the main stem of Model T, makes no difference. However, because the digital numbers are finite in length round-off or truncation errors are introduced at each step of the computation The overall error increases as the number of computational steps increase. Some types of computation such as differentiation tend to increase the error, while other types such as integration tend to decrease the error. For this reason, overall system error is related in a complex way to the order of computation. An understandin of error buildup and testing of the various algorithms will help in choosing the computational sequence that results in smallest errors In the case of cascaded resonators, it is better to perform the computation in reverse order from that implied by Fig. 15 - radiation effect first, then higher order filters in descending center frequency order, then formant filters.

Resonator No.	Center Freq. (Hz)	Bandwidth (Hz)
4	3500	175
5	4500	281
6	5500	458
7	6500	722
8	7500	1250
9	8500	2125
10	9500	4750

Figure 16. Higher Order Correction Filter Center Frequencies and Bandwidths. From Rabiner (1968)

```
C ZERO VARIABLE HOLDING NEXT SPEECH WAVE POINT
      YN=0.0
C ADD GLOTTAL WAVE
      YN=YN+GLOT(P,AV)
C ADD FRICATIVE NOISE
      YN=YN+AN*(FLOAT(IRN4(X))/2047.0)
C APPLY FORMANT FILTERS
      DO 200 I=1,3
200   YN=RES(YN,YM1(I),YM2(I),AO(I),A1(I),A2(I))
C APPLY HIGHER ORDER CORRECTING FILTERS
      DO 250 I=1,7
250   YN=RES(YN,HYM1(I),HYM2(I),HAO(I),HA1(I),HA2(I))
C APPLY RADIANCE EFFECT
      YN=RAD(YN,RYM1,GRAD)
```

Figure 17. Model T Logic in Fortran.

The conversion of the Model T block diagram into Fortran is illustrated by Figure 17, page 27, the series elements being here computed in their natural order. This coding is an example of what should be inserted into the over-all logic (Figure 3, page 9) following the comment lines "GENERATE NEXT SPEECH WAVE POINT...".

B. Control

In the Model T organization, nine control parameters are available -- P (pitch), AV (amplitude of voicing), AN (amplitude of noise) and the center frequency and bandwidth of three variable formant filters. In the main loop, just before generating the next speech wave point, a subroutine (it can be in-line code, of course) calculating values of these parameters is needed. If at the beginning of the program the variable T (time) is initialized to zero and incremented by TDEL at the end of the main loop, it can serve as a simulated-time clock on which to base calculation of the control parameters. The simplest method of control is to formulate the desired control parameter curves algebraically and just

include Fortran statements in this section calculating their values as in the algebraic equations. For example, suppose we wanted the pitch to rise linearly from 80 to 120 Hz in the first 100 msec, stay constant at 120 Hz for 200 msec, then fall linearly to 100 Hz in the next 100 msec and stay at that value from then on. The following Fortran statements can be used to calculate P:

```
      IF (T-100.0) 210,210,220
210   P=80.0+40.0*T/100.0
      GO TO 270
220   IF (T-300.0) 230,230,240
230   P=120.0
      GO TO 270
240   IF (T-400.0) 250,250,260
250   P=120.0-20.0*(T-300.0)/100.0
      GO TO 270
260   P=100.0
270   CONTINUE
```

When new values of CF and BW are computed for the three variable formant filters, subroutine COEFF should be called to translate these into the coefficients A0, A1, and A2 actually used by function RES.

If new values of the control parameters are calculated every sample point, the execution time of the program will be very long There are several obvious ways to speed up this calculation.

One way is to calculate new values for P and AV only at the beginning of each pitch period, since this is the only time GLOT uses them.

With some error introduced, the control parameters can be re-computed only every so many msec to speed things up. A variable used as a time clock in the same way that TG is used by GLOT can control this period. Computing the source control parameters (P, AV, and AN) this way introduces error only in that the actual parameter curves will follow the desired curve in a step-wise fashion, but changing the characteristics of the formant filters this way will introduce another type of error, which comes out sounding like clicks or static if the change in filter characteristics is too large.

Our currently implemented assembly-language synthesizer reads a file of tabled values created by another program as values representing the parameter curves. The period between tabled parameter values is changeable, but on the order of 5 to 10 msec. In computing the actual parameter values used, the synthesizer interpolates linearly along the tabled parameter data curves. The step size of the interpolation can be easily changed, allowing a smooth trade-off between accuracy and execution time.

To sum up, computing new control parameter values for each sample point generated is the easiest and most accurate way, but alternative schemes allowing a convenient trade of accuracy for speed are easily programmed

C. Other Models

We will briefly describe several alternative organizations of the elements, although most of our practical experience has been with the Model T organization.

1. Parallel Formant

One model used in some synthesizers is the parallel formant model, whose block diagram is given as Figure 18, page 32. In serial formant models such as Model T, no independent control of the relative intensities of formants is possible, since the order of operations is immaterial. It has been shown that the relative intensities of formants in a serial synthesis closely match those found in natural speech,[1] which is some justification of the serial model as an analog of the vocal tract. But in a parallel formant arrangement, each parallel channel must have a separate gain control. This is fine if you're investigating the perception of relative formant intensities, but not many have chosen this model for general speech synthesis

Rabiner (1968) contains a worthwhile discussion of the relative merits of serial and parallel synthesis.[2] If you decide to try a parallel formant model, the higher order correcting filters are apparently unnecessary, and Rabiner (1968) mentions that zeros -- anti-resonances -- are introduced into the spectrum.

2. Separate Noise Shaping Channel

In Model T, the same three filters are used to make the formants of voiced speech and to shape the spectrum of noise during unvoiced speech. This is cumbersome and difficult, and a simple alternative is illustrated in Figure 19, page 34.

---

1. Fant (1956)
2. so does Flanagan (1957)

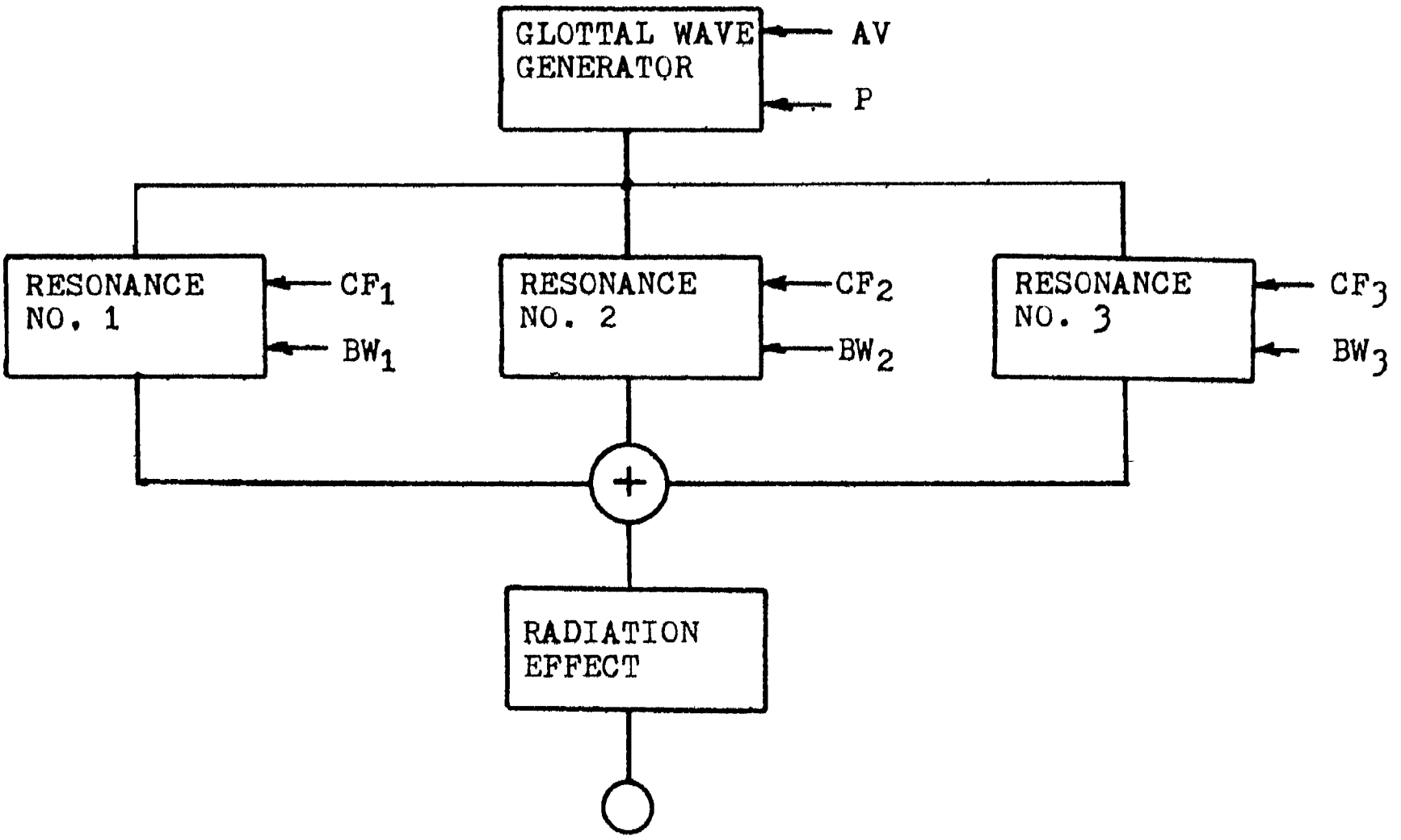

Figure 18. Parallel Formant Organization Model

A separate channel is devoted to noise, with its own resonant and anti-resonant filters for spectral shaping. It has been suggested that one resonance and one anti-resonance are sufficient to model most English fricatives.[1] Of course, this model adds two new control parameters to be computed.

3. Other More Complex Models

Model T does not use anti-resonances; they are not typical of voiced speech, but rather are present in the spectra of fricatives and nasalized segments. For making nasal sounds, a parallel nasal channel whose input is the glottal wave and whose output is added in just before the radiance effect calculation can be added. The spectral shaping filters needed in this channel are not obvious from published reports, but one variable anti-resonance and several fixed resonances are probably a minimum complement.

A multitude of more complex models can be seen in the literature -- Rabiner (1968), to take one example, includes a special arrangement for generating voiced fricatives.

D. A Complete Example

To illustrate the capabilities of the simplest synthesis model, on the following pages we present as Figure 20 a complete Fortran program for synthesizing the word "seat" ([sit^h]). We tried to duplicate one particular token utterance of this word. Spectrograms of the original sound used as a model and the synthesized sound calculated by the Fortran program are shown in

---

1. Heinz and Stevens (1961)

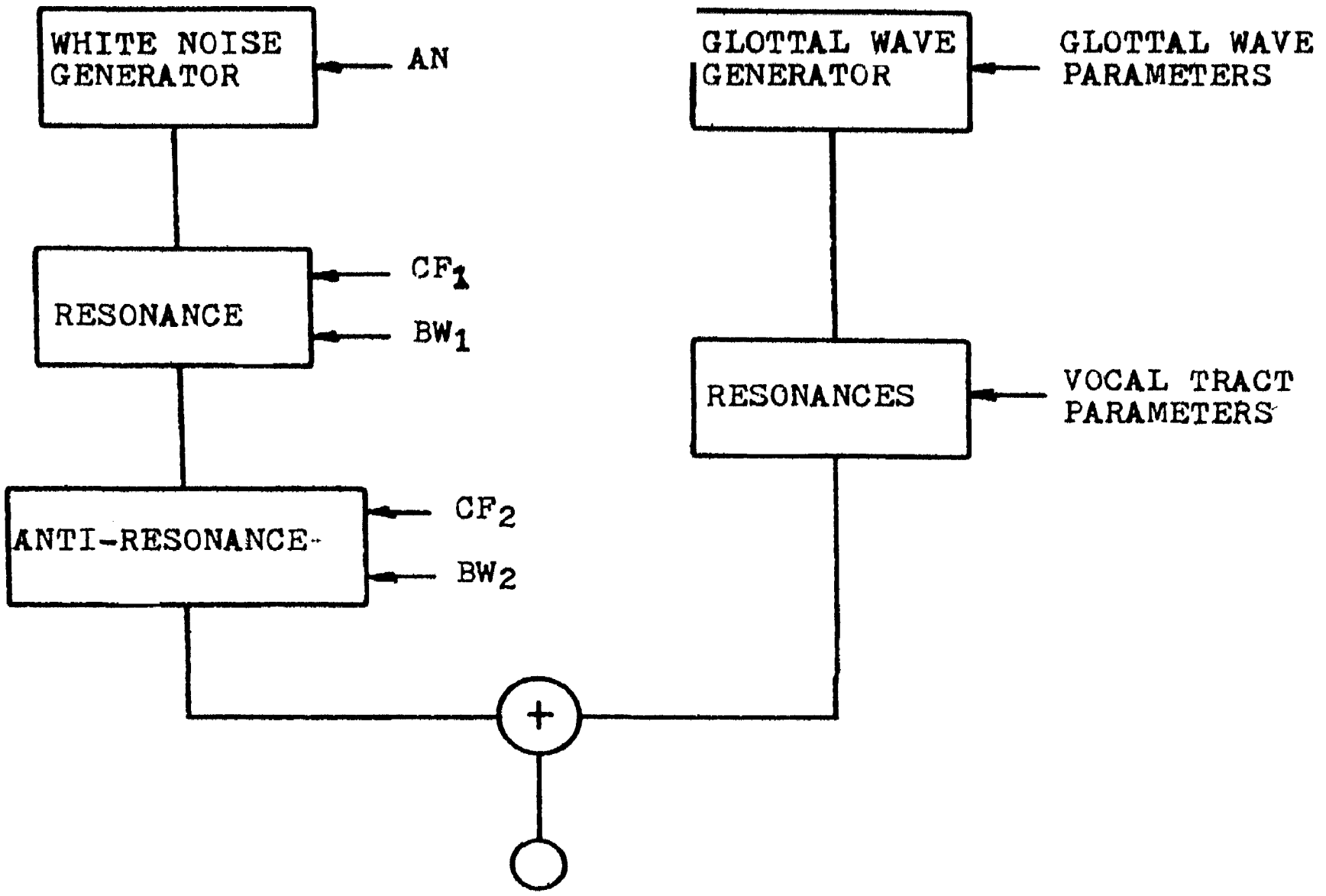

Figure 19. Block Diagram of Model With Separate Noise Shaping Channel

Figure 21, page 44. Deriving the parameter curves from the token utterance took a lot of work, but as Figure 21 shows, the resulting synthetic word is a reasonably close copy of the original.

For those who may want to use this program as a beginning to their work, several features of it will be explained.

The basic model used is the simplest, model "T" (Figure 15, page 25) with one addition: the noise signal is multiplied by a relative gain constant (GFRIC) before entering the vocal tract (variable filter) section.

The output sound wave is stored a block at a time in file RAPWRK1. This file is opened by the subroutine called in line 43 written into in line 455, and closed in line 470.

Parameter values are periodically calculated from piece-wise polynomial algebraic specifications in the section called "GPAR", lines 142 to 415. The periods between parameter re-calculations are controlled by two variables serving as clocks, TVOC for voicing parameters and TFRIC for frication parameters. The values of PVOC and PFRIC are the times in msec between re-calculations of vocalic and fricative parameter values, respectively. These parameter values can be reset more often by merely changing the values assigned to PVOC and PFRIC in lines 33 and 37; at present frication parameters are reset every 0.1 msec and vocalic parameters every 0.2 msec. The Fortran logic calculating the parameters was coded for clarity, not economy, and though lengthy should be easy to follow. The primitive subroutine TPOW returns powers of a variable for ease in calculating polynomial functions of time: after calling TPOW(T,N,TX) $TX(1)=T^1$, $TX(2)=T^2$, ... $TX(N)=T^N$

Certain parameters are constant during some of the sounds, e.g., spectral parameters during "S". As a minor concession to execution speed, these constant parameters are not reset after the first entry into the section during which they are constant The variables ISW1, ISW2, and ISW3 are "first-time-through" switches, used to remember whether or not the temporarily constant parameters have been calculated yet.

We have found that duplicating a token of natural speech using this simple basic synthesis model, though possible, can be quite difficult, requiring much trial-and-error work. Fortunately, much research on speech perception does not require exact duplication of given utterances, but instead uses simpler sets of parameter curves. An example of such a program, which synthesizes the vowel /i/ with constant pitch and intensity can be made by substituting the following code for the "GPAR" Section, lines 151 to 411 of the sample program.

```
      IF (ISW1) 150,100,150
100   ISW1=1
      P=100.0
      AVDB=50.0
      AV=10.0**(AVDB/20.0)
      AN=0.0
      CF(1)=270.0
      CF(2)=2290.0
      CF(3)=3010.0
      BW(1)=54.0
      BW(2)=55.0
      BW(3)=170.0
      DO 110 I=1,3
110   CALL  COEFF(CF(I),BW(I),A0(I),A1(1),A2(I),OSR,1)
150   CONTINUE
```

```
0001        C         FORTRAN SYNTHESIS TESTER
0002        C         WSYN18.FO
0003        C         SYNTHESIZES "SEAT"
0004        C
0005                  COMMON ISWV,DT,TG,TP,T1,T2,OPTR,CLTR,AVSAVE
0006                  DIMENSION IBUF(256),CF(10),BW(10),A0(10),A1(10),A2(10),
0007                  YM1(10),YM2(10),TX(10)
0010        C
0011        C
0012        C         O V E R A L L   I N I T I A L I Z A T I O N
0013        C
0014        C
0015        C SET OUTPUT SAMPLE RATE IN SAMPLES/SEC
0016                    OSR=20000.0
0017        C SET OUTPUT BLOCK SIZE AND NO. OF BLOCKS
0020                    NBSIZE=256
0021                    NBLKS=51
0022        C SET WAVE SHAPE CONSTANTS FOR GLOTTAL WAVE GENERATION
0023                    OPTR=0.40
0024                    CLTR=0.16
0025        C SET OVERALL TIME CLOCK AND DELTA T IN MSEC.
0026                    T=0.0
0027                    DT=1000.0/OSR
0030        C SET PARAMETER=RESETTING GLOCK FOR FRICATION
0031                    TFRIC=0.0
0032        C SET PARAMETER=RESETTING PERIOD FOR FRICATION
0033                    PFRIC=0.1
0034        C SET PARAMETER=RESETTING CLOCK FOR VOICING
0035                    TVOC=0.0
0036        C SET PARAMETER=RESETTING PERIOD F
0037                    PVOC=0.2
0040        C OPEN OUTPUT STORAGE FILE
0041        C OUTPUT BUFFER IS ARRAY IBUF
0042                    NOUT=5
0043                    CALL OPENR(NOUT,"RAPWRK1 ","DD",1,LENG,IBUF,%10)
0044                    GO TO 20
0045        C ERROR HANDLING IF OUTPUT FILE CANNOT BE OPENED
0046        10          WRITE(1,15)
0047        15          FORMAT(" *** ERROR OPENING WORK FILE")
0050                    GO TO 900
0051        C FILE OPENED O.K., CHECK IF BIG ENOUGH
0052        20          IF(NBLKS-LENG) 35,35,25
0053        C ERROR HANDLING -- FILE NOT BIG ENOUGH
0054        25          WRITE(1,30)
0055        30          FORMAT(" *** WORK FILE NOT BIG ENOUGH")
0056                    GO TO 900
0057        C OUTPUT WORK FILE READY TO GO
0060        C SET VOICING SWITCH OFF
0061        35          ISWV=0
0062        C INITIALIZE HIGHER ORDER FILTER VALUES
0063        C IN REVERSE ORDER BY CENTER FREQUENCY
0064                    CF(4)=9500.0
0065                    CF(5)=8500.0
0066                    CF(6)=7500.0
0067                    CF(7)=6500.0
0070                    CF(8)=5500.0
0071                    CF(9)=4500.0
```

Figure 20. Example Program Synthesizing "Seat".

```
0072                CF(10)=3500.0
0073                BW(4)=4750.0
0074                BW(5)=2125.0
0075                BW(6)=1250.0
0076                BW(7)=722.0
0077                BW(8)=458.0
0100                BW(9)=281.0
0101                BW(10)=175.0
0102        C COMPUTE RECURSIVE EQUATION COEFFICIENTS FOR HIGHER ORDER FILTERS
0103                DO 40 I=4,10
0104        40      CALL COEFF(CF(I),BW(I),A0(I),A1(I),A2(I),OSR,1)
0105        C ZERO INITIAL CONDITIONS OF FILTERS (REMEMBERED PAST VALUES)
0106                DO 50 I=1,10
0107                YM1(I)=0.0
0110        50      YM2(I)=0.0
0111        C SET OVERALL GAIN FACTORS
0112                GAIN=1.0/1000.0
0113                GFRIC=1.0/1000000.0
0114        C CLEAR UTILITY SWITCHES
0115                ISW1=0
0116                ISW2=0
0117                ISW3=0
0120        C
0121        C
0122        C       E N D   O F   O V E R A L L   I N I T I A L I Z A T I O N
0123        C
0124        C
0125        C
0126        C
0127        C       M A I N   P R O C E S S I N G   L O O P
0130        C
0131        C
0132        C LOOP ON NO. OF BLOCKS
0133                DO 800 IBLK=1,NBLKS
0134        C LOOP ON OUTPUT POINTS PER BLOCK
0135                DO 750 IPT=1,NBSIZE
0136        C ZERO YN
0137                YN=0.0
0140        C GET NEW PARAMETER VALUES
0141        C
0142        C*****************************************************************
0143        C*                                                               *
0144        C*                                                               *
0145        C*                  GPAR                                         *
0146        C*                                                               *
0147        C* IN-LINE SUBROUTINE TO GET NEW PARAMETER VALUES                 *
0150        C*                                                               *
0151        C DECREMENT PARAMETER-RESETTING CLOCK FOR FRICATION
0152                TFRIC=TFRIC-DT
0153        C DECREMENT PARAMETER-RESETTING CLOCK FOR VOICING
0154                TVOC=TVOC-DT
0155        C "S" ?
0156                IF (T-219,51) 100,200,200
0157        100     CONTINUE
0160        C
0161        C                       "S"
0162        C
```

Figure 20 (continued)

```
0163        C IF FIRST ENTRY INTO "S" ROUTINE, SET CONSTANT PARAMETERS
0164              IF (ISW1) 105,105,115
0165        105   AV=0.0
0166              CF(1)=4000.0
0167              CF(2)=5700.0
0170              CF(3)=9000.0
0171              BW(1)=1500.0
0172              BW(2)=3000.0
0173              BW(3)=4000.0
0174              DO 110 I=1,3
0175        110   CALL COEFF(CF(I),BW(I),A0(I),A1(I),A2(I),OSR,1)
0176              ISW1=1
0177        C IF TIME TO DO SB, RESET VARIABLE PARAMETERS
0200        115   IF (TFRIC) 120,120,125
0201        120   CONTINUE
0202        C IF T<5, NO NOISE
0203              IF (T-5.0) 122,124,124
0204        122   AN=0.0
0205              GO TO 125
0206        124   CALL TPOW(T,4,TX)
0207              ANDB=+3.50+1.6375*TX(1)-0.02032*TX(2)+0.0001114*TX(3)
0210           1  -0.0000002242*TX(4)
0211              AN=10.0**(ANDB/20.0)
0212        125   CONTINUE
0213              GO TO 650
0214        C
0215        C                              END OF "S"
0216        C
0217        C "EA" VOWEL?
0220        200   IF (T-440.90) 205,310,310
0221        C
0222        C                               "EA"
0223        C
0224        C IF FIRST TIME THRU, SET CONSTANT PARAMETERS
0225        205   IF (ISW2) 210,210,215
0226        210   AN=0.0
0227              BW(1)=54.0
0230              BW(2)=55.0
0231              BW(3)=170.0
0232              ISW2=1
0233        C IF TIME, RESET VARIABLE PARAMETERS
0234        215   IF (TVOC) 220,220,650
0235        220   CONTINUE
0236        C PITCH
0237              CALL TPOW(T-330.0,4,TX)
0240              P=146.243-0.4251854*TX(1)-0.0001925078*TX(2)
0241           1  -0.000005018408*TX(3)-0.00000062460681*TX(4)
0242        C AMPLITUDE OF VOICING
0243              IF (T-245.0) 410,410,420
0244        410   CALL TPOW(T-219.51,3,TX)
0245              AVDB=43.493+0.619573*TX(1)-0.02476343*TX(2)
0246           1  +0.0002630669*TX(3)
0247              GO TO 510
0250        420   IF (T-290.0) 430,430,440
0251        430   AVDB=47.34763-0.1457143*(T-245.0)
0252              GO TO 510
0253        440   IF(T-335.0) 450,450,460
```

Figure 20. (continued)

```
0254        450       AVDB=40.8-0.072*(T-290.0)
0255                  GO TO 510
0256        460       IF (T-400.0) 470,470,480
0257        470       AVDB=37.29645+0.045*(T-335.0)
0260                  GO TO 510
0261        480       IF (T-420.0) 490,490,500
0262        490       CALL TPOW(T-400.0,2,TX)
0263                  AVDB=39.86422-0.07127*TX(1)-0.02243*TX(2)
0264                  GO TO 510
0265        500       CALL TPOW(T-420.0,2,TX)
0266                  AVDB=29.48846-1.379416*TX(1)+0.02857102*TX(2)
0267        510       AV=10.0**(AVDB/20.0)
0270        C CF1
0271                  IF (T-260.0) 225,226,226
0272        225       CF(1)=250.0+1.23*(T-219.51)
0273                  GO TO 230
0274        226       IF (T-395.0) 227,228,228
0275        227       CF(1)=300.0
0276                  GO TO 230
0277        228       CF(1)=300.0-4.8*(T-395.0)
0300        230       CONTINUE
0301        C CF(2)
0302                  IF (T-320.0) 232,232,234
0303        232       CALL TPOW(T-219.51,2,TX)
0304                  CF(2)=1732.857+14.95833*TX(1)-0.071176*TX(2)
0305                  GO TO 250
0306        234       IF (T-395.0) 235,235,240
0307        235       CF(2)=2500.0
0310                  GO TO 250
0311        240       CF(2)=2500.0-20.0*(T-395.0)
0312        250       CONTINUE
0313        C BW(2) IS CONSTANT
0314        C CF(3)
0315                  IF (T-344.0) 255,255,260
0316        255       CALL TPOW(T-219.51,2,TX)
0317                  CF(3)=2411.076+10.77844*TX(1)-0.04028484*TX(2)
0320                  GO TO 275
0321        260       IF (T-395.0) 265,265,270
0322        265       CF(3)=3125.0
0323                  GO TO 275
0324        270       CF(3)=3125.0-25.0*(T-395.0)
0325        275       CONTINUE
0326        C BW(3) IS CONSTANT
0327        300       CONTINUE
0330        C NO NEED TO RESET FILTERS IF 350<T<394
0331                  IF (T-350.0) 305,302,302
0332        302       IF (T-394.0) 650,305,305
0333        305       DO 308 I=1,3
0334        308       CALL COEFF(CF(I),BW(I),A0(I),A1(I),A2(I),OSR,1)
0335                  GO TO 650
0336        C
0337        C
0340        C                    SILENCE+RELEASE OF "T"
0341        C
0342        310       IF (T-502.70) 315,320,320
0343        315       AN=0.0
0344                  AV=0.0
```

Figure 20. (continued)

```
0345                GO TO 400
0346        320     IF (T-617.77) 325,315,315
0347        C IF FIRST TIME, SET CONSTANT FEATURES OF RELEASE
0350        325     IF (ISW3) 330,330,340
0351        330     ISW3=1
0352                CF(1)=1800.0
0353                CF(2)=1800.0
0354                CF(3)=4300.0
0355                BW(1)=150.0
0356                BW(2)=150.0
0357                BW(3)=2500.0
0360                DO 335 I=1,3
0361        335     CALL COEFF(CF(I),BW(I),A0(I),A1(I),A2(I),OSR,1)
0362        C AN
0363        340     IF (TFRIC) 345,345,400
0364        345     IF (T-502.81) 346,347,347
0365        346     ANDB=111.0
0366                GO TO 395
0367        347     IF (T-512.0) 350,355,355
0370        350     ANDB=94.5
0371                GO TO 395
0372        355     IF (T-538.0) 360,365,365
0373        360     ANDB=89.5
0374                GO TO 395
0375        365     ANDB=89.5-0.40*(T-538.0)
0376        395     AN=10.0**(ANDB/20.0)
0377        400     CONTINUE
0400        C
0401        C
0402        C
0403        C FINAL STEP -- RESET TIMERS
0404        C
0405        650     IF (TFRIC) 660,660,670
0406        660     TFRIC=PFRIC
0407        670     IF (TVOC) 680,680,690
0410        680     TVOC=PVOC
0411        690     CONTINUE
0412        C*                                                              *
0413        C*      END OF GPAR                                             *
0414        C*                                                              *
0415        C****************************************************************
0416        C
0417        C ADD GLOTTAL WAVE
0420                YN=YN+GLOT(P,AV)
0421        C ADD NOISE
0422        C MULTIPLIED BY RELATIVE NOISE GAIN
0423                YN=YN+GFRIC*AN*FLOAT(IRN4(X))/2047.0
0424        C APPLY LOWER THREE (FORMANT) FILTERS
0425        C AND HIGHER ORDER CORRECTION FILTERS
0426                DO 700 I=1,10
0427        700     YN=RES(YN,YM1(I),YM2(I),A0(I),A1(I),A2(I))
0430        C SIMULATE RADIATION EFFECT
0431                YN=RAD(YN,YM1RAD,OSR)
0432        C MULTIPLY BY OVERALL GAIN FACTOR
0433                YN=YN*GAIN
0434        C CHECK FOR CLIPPING
0435                IF (ABS(YN)-2047.0) 730,730,710
```

Figure 20. (continued)

```
0436        C ERROR -- SPEECH IS CLIPPED
0437        710     WRITE(1,720)
0440        720     FORMAT(" *** CLIPPED")
0441                GO TO 900
0442        C O.K. -- NOT CLIPPED
0443        730     CONTINUE
0444        C STORE SPEECH WAVE POINT
0445                IBUF(IPT)=IFIX(YN)
0446        C INCREMENT GENERAL TIME CLOCK
0447                T=T+DT
0450        750     CONTINUE
0451        C
0452        C       END OF BLOCK-SIZED LOOP
0453        C
0454        C WRITE OUT BLOCK
0455                CALL WRITR(NOUT,IBLK,IBUF,%760)
0456                GO TO 800
0457        C I/O ERROR HANDLING
0460        760     WRITE(1,770)IBLK
0461        770     FORMAT(" *** I/O ERROR BLOCK NO. ",I5)
0462                GO TO 900
0463        800     CONTINUE
0464        C
0465        C       END OF LOOP ON NO. OF BLOCKS
0466        C
0467        C CLOSE WORK FILE
0470                CALL CLOSR(NOUT)
0471        C END
0472        900     WRITE(1,910)
0473        910     FORMAT(" HIT SPACE BAR TO QUIT")
0474                PAUSE
0475                CALL EXIT
0476                END
0477        C
0500        C WSYN1.FO
0501        C
0502        C SYNTHESIZES "SEAT"
0503        C 9/27/74
```

Figure 20. (continued)

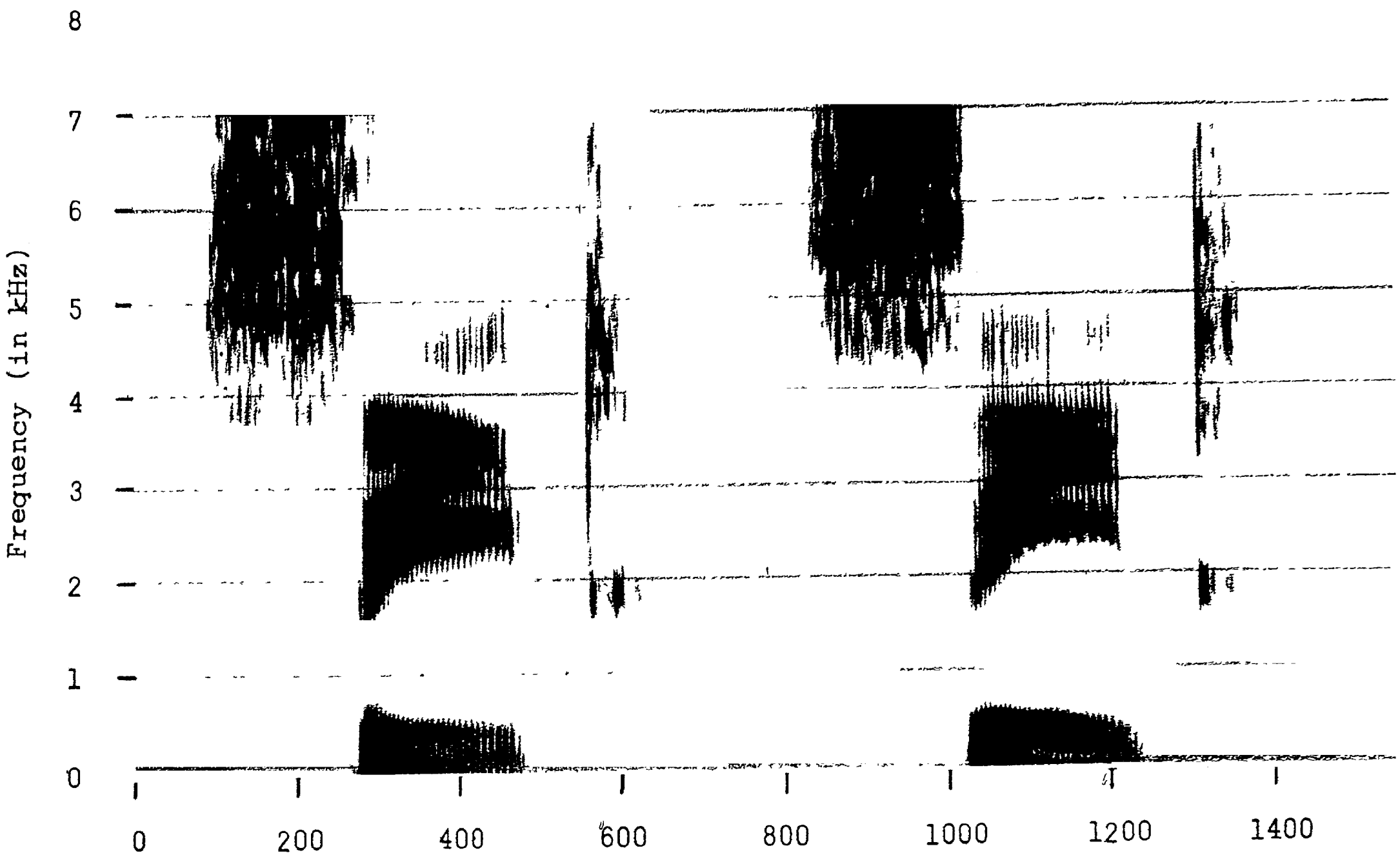

Figure 21. Spectrographic Comparison of Natural and Synthesized "Seat". This spectrogram was made on the Voice Identification, Inc. Series 700 machine with wide-band setting and high frequency shaping above 1 kHz.

V. Final Remarks

The simple schemes for speech synthesis presented here are sufficient to make speech containing most of the sounds of English.

Figure 22, page 45, is a handy reference containing characteristics of English vowels from several primary sources, assembled in tabular form by Dunn as part of his section of Automatic Speech Recognition (1963). Since this reference is not easily available, the primary sources of Dunn's data are also listed in the bibliography. The fundamental and three formant frequencies for men, women, and children were taken from Peterson and Barney (1952). The three formant amplitudes for each vowel are also from the Peterson and Barney measurements, but averaged over all three classes of speakers. Dunn took the relative strengths of the first formants of the different vowels from the measurements of Sacia and Beck (1926), as quoted by Fletcher (1953) arbitrarily assigning the zero decibel level to the strongest vowel, [ɔ] The three formant bandwidths are the average of three sets given by Fant (1962), one set his own measurements, one those of House and Stevens, and the third measured by Dunn himself (1961). Dunn cautions that the bandwidths vary widely among individual speakers.

These values of center frequency and bandwidth of formants for the different vowels can be used as variable formant filter control parameters in Model T with unimpeachable results.

Vowel	i	ɪ	ɛ	æ	ɑ	ɔ	u	ʊ	ʌ	ɜ
As in	heed	hid	head	had	hod	hawed	who'd	hood	hud	heard
Fundamental Frequencies (cycles per second)										
M	136	135	130	127	124	129	141	137	130	133
W	235	232	223	210	212	216	231	232	221	218
Ch.	272	269	260	251	256	263	274	276	261	261
Formant Frequencies (cycles per second)										
F_1 M	270	390	530	660	730	570	300	440	640	490
W	310	430	610	860	850	590	370	470	760	500
Ch.	370	530	690	1010	1030	680	430	560	850	560
F_2 M	2290	1990	1840	1720	1090	840	870	1020	1190	1350
W	2790	2480	2330	2050	1220	920	950	1160	1400	1640
Ch.	3200	2730	2610	2320	1370	1060	1170	1410	1590	1820
F_3 M	3010	2550	2480	2410	2440	2410	2240	2240	2390	1690
W	3310	3070	2990	2850	2810	2710	2670	2680	2780	1960
Ch.	3730	3600	3570	3320	3170	3180	3260	3310	3360	2160
Formant Amplitudes (db)										
L_1	-4	-3	-2	-1	-1	0	-3	-1	-1	-5
L_2	-24	-23	-17	-12	-5	-7	-19	-12	-10	-15
L_3	-28	-27	-24	-22	-28	-34	-43	-34	-27	-20
Formant Bandwidths (cycles per second)										
B_1	54	53	48	63	54	43	50	40	52	44
B_2	55	69	69	81	57	47	49	44	57	58
B_3	170	113	101	126	93	68	77	62	89	64

Figure 22. Average Measured Characteristics of Vowels From Dunn's section (p. D-2) of Automatic Speech Recognition (1963).

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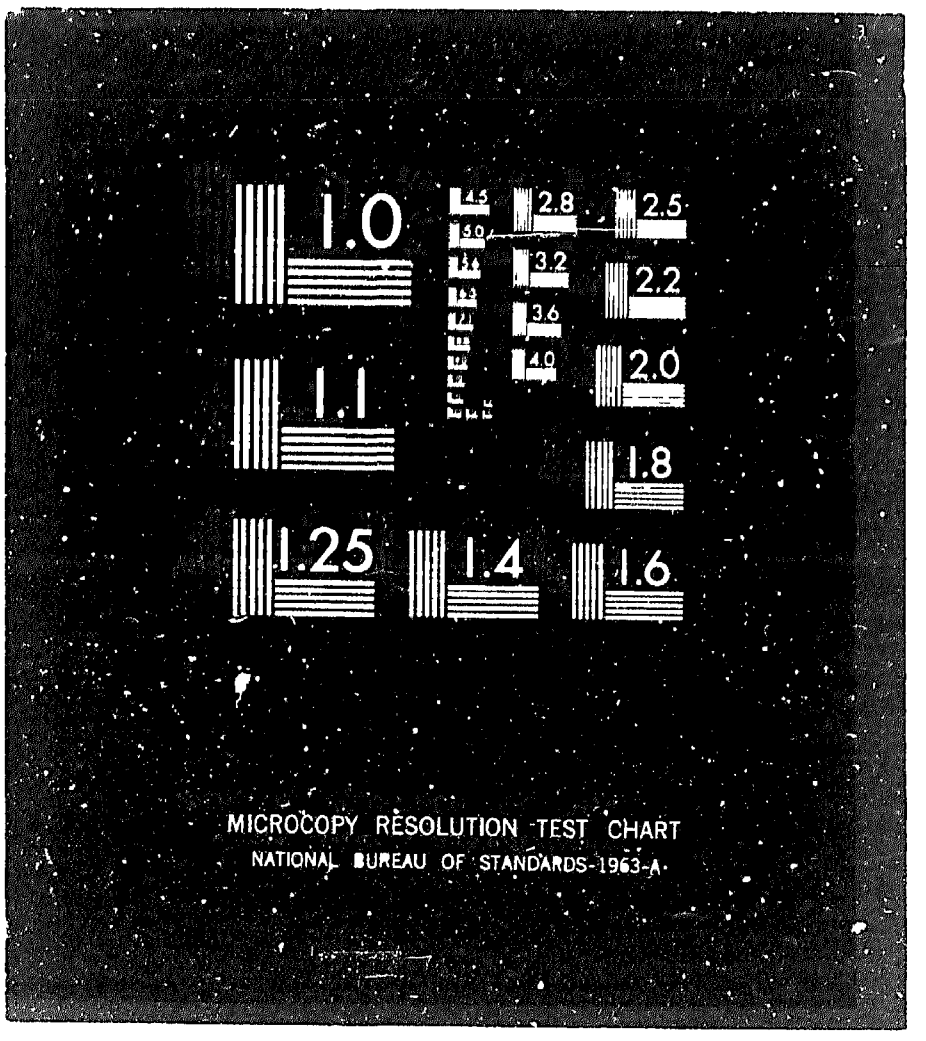
An errata sheet for the article by Lovell et al. is available from John D. Lovell, Dept. of Psychology, California State College, Hayward, CA 94542.

END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MICROCOPY RESOLUTION TEST CHART
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