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FORTRAN IMPLEMENTATION OF COMPLEX LEAST SQUARES
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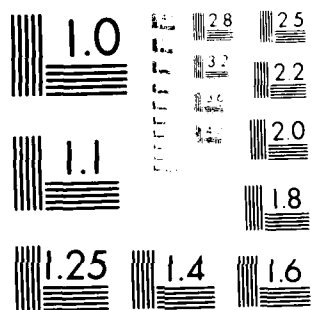
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FORTRAN IMPLEMENTATION OF COMPLEX LEAST SQUARES
ADAPTIVE LATTICE ALGORITHM AS AN ALL ZERO INVERSE
FILTER

B. A. Cooper and L. H. Sibul

Technical Memorandum
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ABSTRACT

A complex least squares adaptive lattice algorithm (1) has been implemented on the ARL VAX/VMS computer as an inverse all zero filter. A FORTRAN program package is presented which also includes a recursive realization of a pole-zero prefilter and an input signal generator. A series of examples is taken from the speech field, using the inverse filter to deconvolve speech signals produced by vocal tract models of varying complexity. The input to the prefilter is shown to be accurately recovered as the output of the inverse filter.

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INTRODUCTION

A complex least squares lattice adaptive algorithm (1) has been implemented on the ARL VAX/VMS computer as an inverse all zero filter. An interactive FORTRAN program package is presented which also includes a recursive realization of a pole-zero prefilter and an input signal generator. No knowledge of FORTRAN is required to use the package. The input signal characteristics, filter coefficients and critical lattice parameters are all user specified.

Three classes of adaptive algorithms are treated in the literature: (Widrow's) LMS, the gradient lattice, and the least squares lattice. Of the three, the least squares adaptive lattice algorithm has been shown to have the fastest convergence rate (2,3). For this reason, it has been used successfully in a wide range of applications; i.e., data deconvolution, system identification, speech processing, and spectral whitening. The program package described in this memorandum was developed with these applications in mind.

BACKGROUND

Consider a filter with transfer function

$$H(z) = \frac{Z(z)}{V(z)} = \frac{B(z)}{A(z)} \quad (1)$$

where

$$A(z) = 1 + \sum_{k=1}^{NPOLE} A_k z^{-k} = \sum_{k=0}^{NPOLE} A_k z^{-k}, \quad A_0 = 1$$

and

$$B(z) = 1 + \sum_{k=1}^{NZERO} B_k z^{-k} = \sum_{k=0}^{NZERO} B_k z^{-k}, B_0 = 1$$

then $Z(z)A(z) = B(z)V(z)$ and, after taking inverse z transforms of both sides, the filter can be realized in the time domain by the following recursion:

$$z(n) + \sum_{k=1}^{NPOLE} A_k z(n-k) = v(n) + \sum_{k=1}^{NZERO} B_k v(n-k)$$

or

$$z(n) = - \sum_{k=1}^{NPOLE} A_k z(n-k) + v(n) + \sum_{k=1}^{NZERO} B_k v(n-k).$$

The roots of $A(z)$ are the poles of the system; the roots of $B(z)$ are the zeroes of the system. For stability, these roots must lie outside the unit circle because they are roots of polynomials in z^{-1} .

The resonances and antiresonances of the human vocal tract (for a particular steady state vowel sound) can be modeled by poles and zeroes, respectively, in the z plane. It is a common practice in the speech field to model these formants with an all pole model rather than with poles and zeroes; adding a sufficient number of poles has been shown to closely approximate the effect of the zeroes. The roots of the denominator polynomial are related to the vocal tract formants by the following expression (5):

$$z_k, z_k^* = z_{2k}, z_{2k-1} = e^{-\sigma_k T} e^{\pm j 2\pi F_k T}$$

where F_k is the formant frequency and $2\sigma_k$ is the bandwidth of the k^{th} formant. There will be a pair of complex conjugate poles for each formant frequency; the order of the system will be twice the number of formants.

Then

$$H(z) = \frac{1}{\prod_{k=1}^{NPOLE} (1 + z_k z^{-1})}.$$

For the all-pole model, the time domain recursion becomes

$$z(n) = -\sum_{k=1}^{NPOLE} A_k z(n-k) + v(n).$$

An inverse filter for the system of Equation 1 would have the transfer function

$$H(z) = \frac{A(z)}{B(z)}.$$

For the speech case (all-pole prefilter), the inverse filter would be all-zero:

$$H_{\text{pre}}(z) = \frac{1}{A(z)}; H_{\text{inv}}(z) = \frac{1}{H_{\text{pre}}(z)} = A(z).$$

Obviously, if the systems are cascaded, the input to the prefilter will be recovered as the output of the inverse filter.

A complex adaptive lattice structure has been used to develop the inverse filter. The actual lattice algorithm is taken from (1). Users are referred to (2) and (3) and the references cited therein for the derivations of the complex least squares adaptive lattice algorithm.

PROGRAM PACKAGE

The lattice program package consists of the following interactive
FORTRAN programs:

GEN.FOR	input generator
FILTER.FOR	prefilter and realization of the lattice filter
LSL.FOR	complex adaptive lattice processor

the following subroutines:

SYNTH	synthesizes complex polynomial from formants and bandwidths
SYNTH1	synthesizes complex polynomial from its roots
ZCPOLY	solves polynomial with complex coefficients

and the following functions:

MTH\$RANDOM	(VAX/VMS library) uniform random number generator
AMAXLOC	finds location of highest value in an array
ARRMAX	finds highest value in an array

The execution of the programs may be controlled by either one of
two command procedures: LATTICE.COM when using a VT100 terminal, or
LATTEK.COM when using the Tektronix terminal in Room 358, ASB. These
command procedures are given in Appendix A. They may be invoked by
typing @LATTICE and @LATTEK, respectively. The following command should
be placed into either the command procedure or the user's LOGIN.COM file
to allow access to the IMSL package. If LNK\$LIBRARY has already been
assigned, use LNK\$LIBRARY1 or the next highest unassigned library.

\$ASSIGN DRA0:[SYSTEM]IMSLIBS.OLB LNK\$LIBRARY

Appendix B presents a sample terminal session for the case illustrated in Figure 3. For the user interested in modifying the programs, the rest of this memorandum deals with the structure of the individual programs.

Figure 7 illustrates the calling hierarchy of the lattice package. The inputs to program LSL are Gaussian white noise and the output time series $z(n)$ from program FILTER.FOR. The outputs from LSL are the set of inverse filter coefficients (A_k ; $k=1, \dots, \text{NZERO}$) to which the lattice has adapted after the requested number of iterations, and the time series output $e_p(n)$ which is equivalent to the result of passing the input time series $z(n)$ through the transfer function $H(z) = A(z)$. The output filter coefficients are fed back into program FILTER to be realized recursively with input $z(n)$. Thus, since program FILTER operates for any $H(z)$, it is both a prefilter and the realization of the lattice filter.

A sample case, then, may be illustrated by a series of five plots (Figures 3-6). The relationships between the plots are shown by the block diagram in Figure 2:

- (a) input time series
- (b) prefilter transfer function
- (c) output of the prefilter
- (d) lattice (inverse) filter transfer function
(realized by program FILTER)
- (e) output of the lattice filter

So that the filter outputs may be most easily evaluated, the filter outputs (c) and (e) for deterministic prefilter inputs are time series plots; for a stochastic input, the outputs (c) and (e) are power spectra. The time series plots (c) and (e) (from above) for the deterministic case are

presented in Figures 3-6 for one period of the waveform after the time series has stabilized (or adapted). Again, the output of the lattice filter should approximate the input to the prefilter if the lattice is a true inverse filter. It is obvious that some measure of goodness of the inverse filter should be applied to the output of the lattice filter. At present, only a graphical comparison has been attempted. However, Gray and Markel (7) suggest a spectral flatness measure which will help to smooth the discontinuities in the power spectrum caused by the occurrence of the inverse filter zeroes being offset by a few Hz. from the prefilter poles. Also, a mean square error criterion might be applied to the difference between the deterministic time series output (error signal) and the input time series. The transfer function of the lattice filter should be the inverse of the prefilter transfer function. Thus, where the prefilter has poles (peaks) the inverse filter will have zeroes (dips).

GEN.FOR

Program GEN.FOR generates both Gaussian white noise and an impulse sequence. A more accurate approximation to the glottal wave is given in (7). However, since the thrust of this work is aimed at demonstrating that the lattice filter is indeed an accurate inverse filter, the two cases used so far are sufficient models of voiced and unvoiced steady state vocal tract excitations.

Program GEN.FOR calls the uniform random number generator, MTHSRANDOM, from the VAX/VMS library. The Gaussian noise is then developed following a technique from (9). The output time series are in complex format but the imaginary parts are zero. The sampling frequency is 10 kHz. For

the complex impulse input, the frequency is variable; the examples shown in Figures 3 and 5 use $F_0 = 125$ Hz, a reasonable fundamental frequency for an adult male voice. The interactive user inputs to program GEN are F_0 , fundamental frequency; M , number of desired samples; and σ , the standard deviation of the noise. The time series are each stored in an output disk file.

FILTER.FOR

The user selects the desired type of input and the source of the filter coefficients. Several options exist concerning specification of coefficients. The formant frequencies and their bandwidths may be specified and the coefficients calculated from subroutine SYNTH and then stored on disk. The stored coefficients may be used with another input option, reading in coefficients from disk. The coefficients may be typed in directly from the keyboard. For the realization of the lattice (inverse) filter, the forward predictor (outputs from program LSL) coefficients may be read in from disk as the lattice filter coefficients. The other interactive user inputs to FILTER.FOR are M , desired number of time samples; $NPOLE$, number of poles; and $NZERO$, the number of zeroes. It is possible to specify a prefilter transfer function which has both poles and zeroes (the inverse filter would still be all zero). The modeling of a pole zero process by an all-pole process is discussed by Friedlander and Maitra (6). The examples presented here involve prefilters containing only poles for the purpose of illustrating the accuracy of the all-zero inverse filter.

The filter gain is computed by the following formula:

$$G = 10 \text{ LOG } \frac{\sum_{n=1}^M z^2(n)}{\sum_{n=1}^M v^2(n)} = \frac{\sigma_{out}^2}{\sigma_{in}^2}$$

where $u = 0$ and $\sigma_z^2 = \sigma_x^2 + \sigma_y^2$.

The filter transfer function is realized from the filter coefficients by the following formula:

$$20 \text{ LOG } |H(\omega)| = 20 \text{ LOG } \left| \frac{1}{A(z)} \right| = 20 \text{ LOG } \left[\frac{1}{A(z)} \cdot \frac{1}{A^*(z)} \right]^{\frac{1}{2}}$$

for the prefilter and

$$20 \text{ LOG } |A(z)| = 20 \text{ LOG } (A(z) \cdot A^*(z))^{\frac{1}{2}}$$

for the lattice filter, where

$$A(z) = 1 + \sum_{k=1}^{\text{NPOLE}} A_k e^{-j\omega k}$$

The frequency range of the plots is 0-5000 Hz.

The output power spectrum is computed (4) for the stochastic case from $10 \text{ LOG } P(\omega)$ where

$$P(\omega) = P_{in}(\omega) \cdot |H(\omega)|^2 = P_{in}(\omega) \cdot \frac{1}{A(z) \cdot A^*(z)}$$

For the prefilter, $P_{in}(\omega) = \sigma_{in}^2 = \sigma_v^2$. For the lattice filter, $P_{in}(\omega)$ will be $P_{out}(\omega)$ from the prefilter, and $P(\omega) = P_{in}(\omega) \cdot |H(\omega)|^2 = P_{in}(\omega) \cdot A(z) \cdot A^*(z)$. Again, the frequency range of the plots is 0-5000 Hz.

LSL.FOR

Figure 1 shows the structure of the least squares complex adaptive lattice. Table I lists the lattice parameters and their relationships to the text equations in (3) and to the FORTRAN code in (1). The algorithm from (1) has been modified (from RATFOR) for use by the ARL VAX/VMS computer. Users are referred to (2) for a detailed explanation of filter parameters. The interactive user inputs to program LSL are SNR, desired signal to noise ratio in dB; ALSL, value of α_{CLSL} ; P, order of the filter; and LIM, the desired number of iterations. After LIM number of iterations, the forward predictor coefficients become the filter coefficients of the inverse filter. The convergence properties of this lattice structure have been studied by Hodgkiss (2,3) and are not considered here. The maximum allowable number of iterations is set at 5000 which allows the lattice more than enough time to adapt.

Subroutine ZCPOLY solves the filter polynomial to make sure the roots are outside the unit circle. If a root is inside the unit circle, it is inverted and a new polynomial is synthesized with subroutine SYNTH1. The coefficients are then stored on disk for use by program FILTER.

The forward and backward reflection (PARCOR) coefficients are also stored on disk for possible future use. These parameters are related to the reflection coefficients of an acoustic tube model of the vocal tract. The next step in this research effort will involve making use of these coefficients to directly identify the prefilter transfer function. Table II lists the prefilter coefficients, final lattice coefficients, and both sets of final PARCOR coefficients for the four sample cases presented in Figures 3-6.

The desired signal to noise ratio is achieved by scaling down the added noise (of user specified standard deviation from GEN.FOR) rather than scaling up the signal. This is accomplished by solving for the desired σ of the noise (8):

$$SNR = 10 \log \frac{\lim_{n=1} \sum \text{sig}^2(n)}{\lim_{n=1} \sum \text{noise}^2(n)}$$

taking the antilog:

$$10^{(SNR/10)} = \frac{\lim_{n=1} \sum \text{sig}^2(n)}{\lim_{n=1} \sum \text{noise}^2(n)}$$

$$\lim_{n=1} \sum \text{noise}^2(n) = \frac{\lim_{n=1} \sum \text{sig}^2(n)}{10^{(SNR/10)}}$$

$$\frac{\lim_{n=1} \sum \text{noise}^2(n)}{\lim} = \sigma_{\text{noise}}^2 = \frac{\lim_{n=1} \sum \text{sig}^2(n)}{\lim \cdot 10^{(SNR/10)}}$$

$$\sigma_{\text{noise}} = \left(\frac{\lim_{n=1} \sum \text{sig}^2(n)}{\lim \cdot 10^{(SNR/10)}} \right)^{\frac{1}{2}}$$

Therefore, multiply the noise samples by the factor $\frac{\sigma_{\text{noise}}}{\sigma_{\text{input}}}$.

ACKNOWLEDGMENTS

The complex least squares lattice algorithm in program LSL is an adaptation of a program from the package developed by W. S. Hodgkiss, D. Alexandrou, and J. A. Presley (1-3,9).

The authors would also like to thank J. R. Sacha for the benefit of his generous help and limitless expertise.

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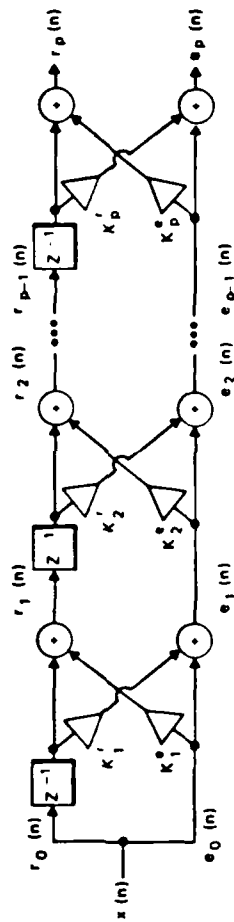
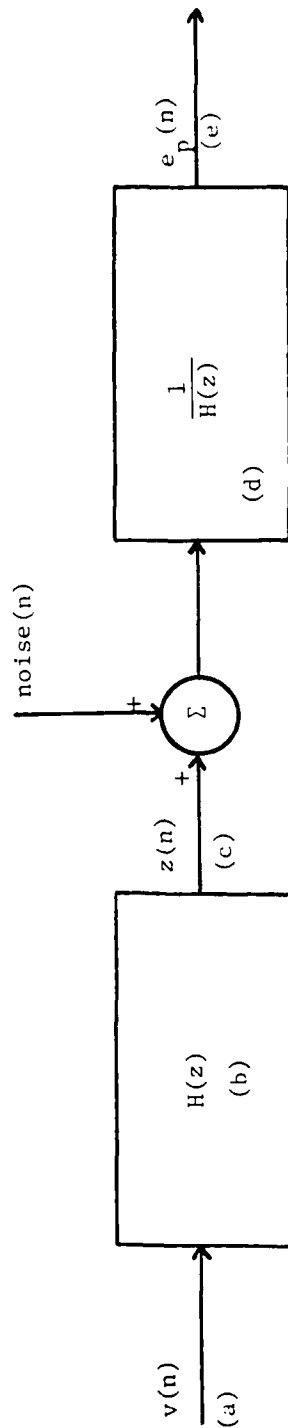


FIGURE 1. LATTICE (INVERSE) FILTER STRUCTURE



- (a) Input to the prefilter, $v(n)$ (impulse sequence or white noise sequence).
- (b) Prefilter transfer function, $20 \log |H(z)|$.
- (c) Output of the prefilter, $z(n)$.
- (d) Lattice (inverse) filter transfer function, $20 \log \left| \frac{1}{H(z)} \right|$.
- (e) Output of the lattice filter (error signal, $e_p(n)$, for deterministic case; output power spectrum, $10 \log P(\omega)$, for stochastic case).

FIGURE 2. BLOCK FLOW DIAGRAM OF LATTICE (INVERSE) FILTER IMPLEMENTATION

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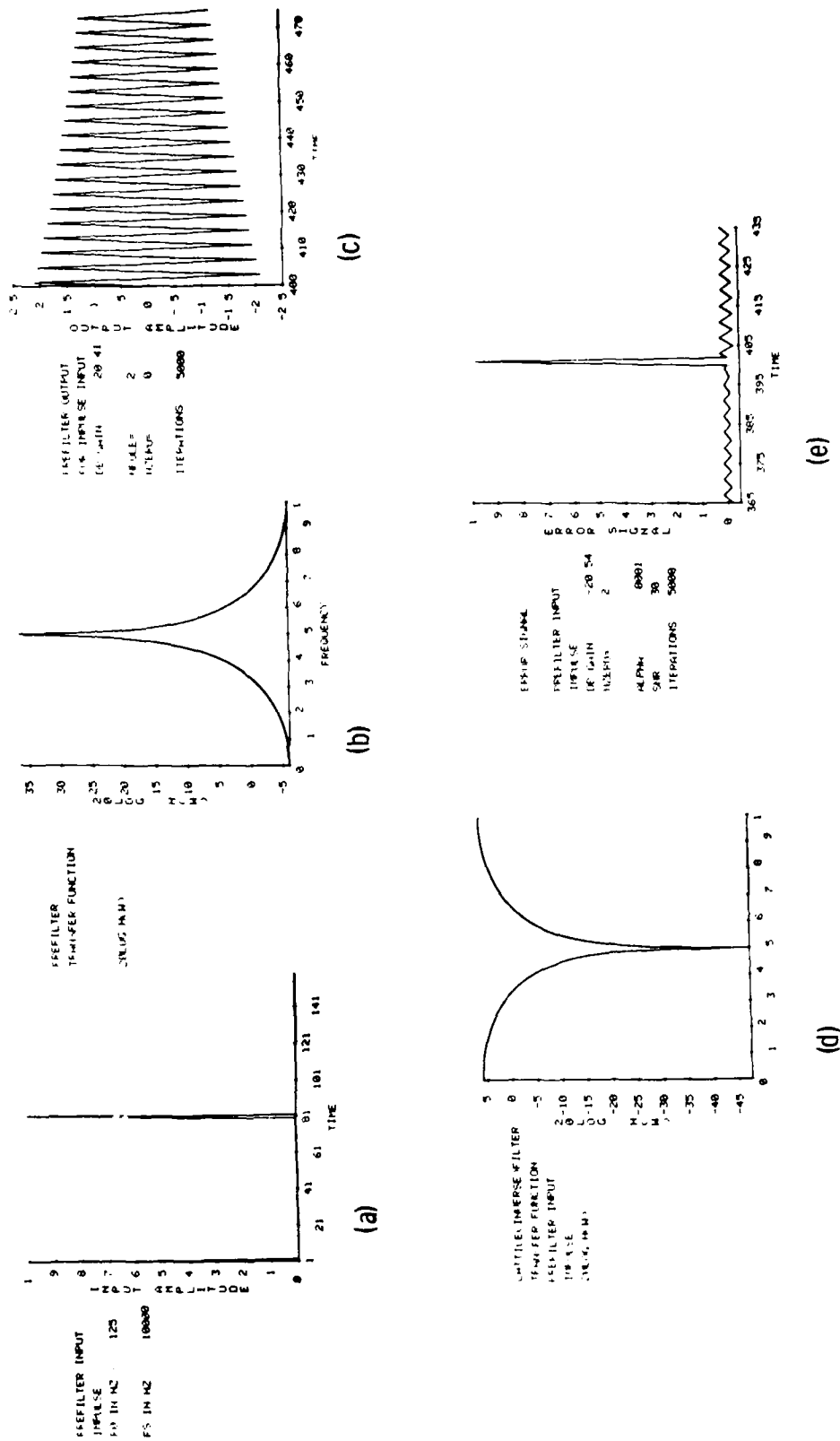
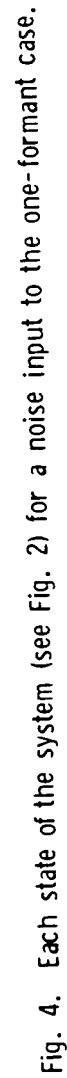


Fig. 3. Each state of the system (see Fig. 2) for an impulse function input to the one-formant case.



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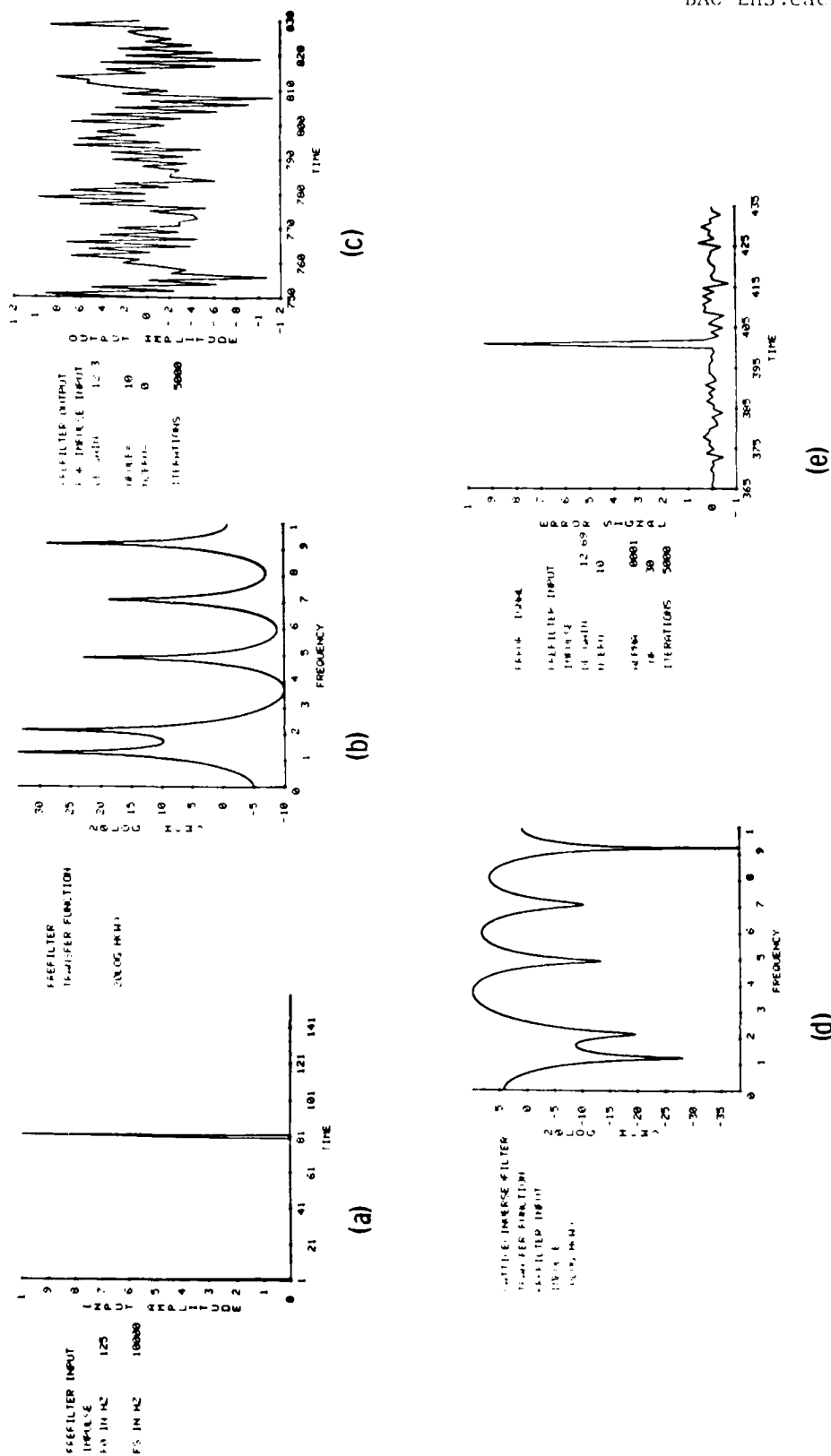


Fig. 5. Each state of the system (see Fig. 2) for an impulse function input to the five-formant case.

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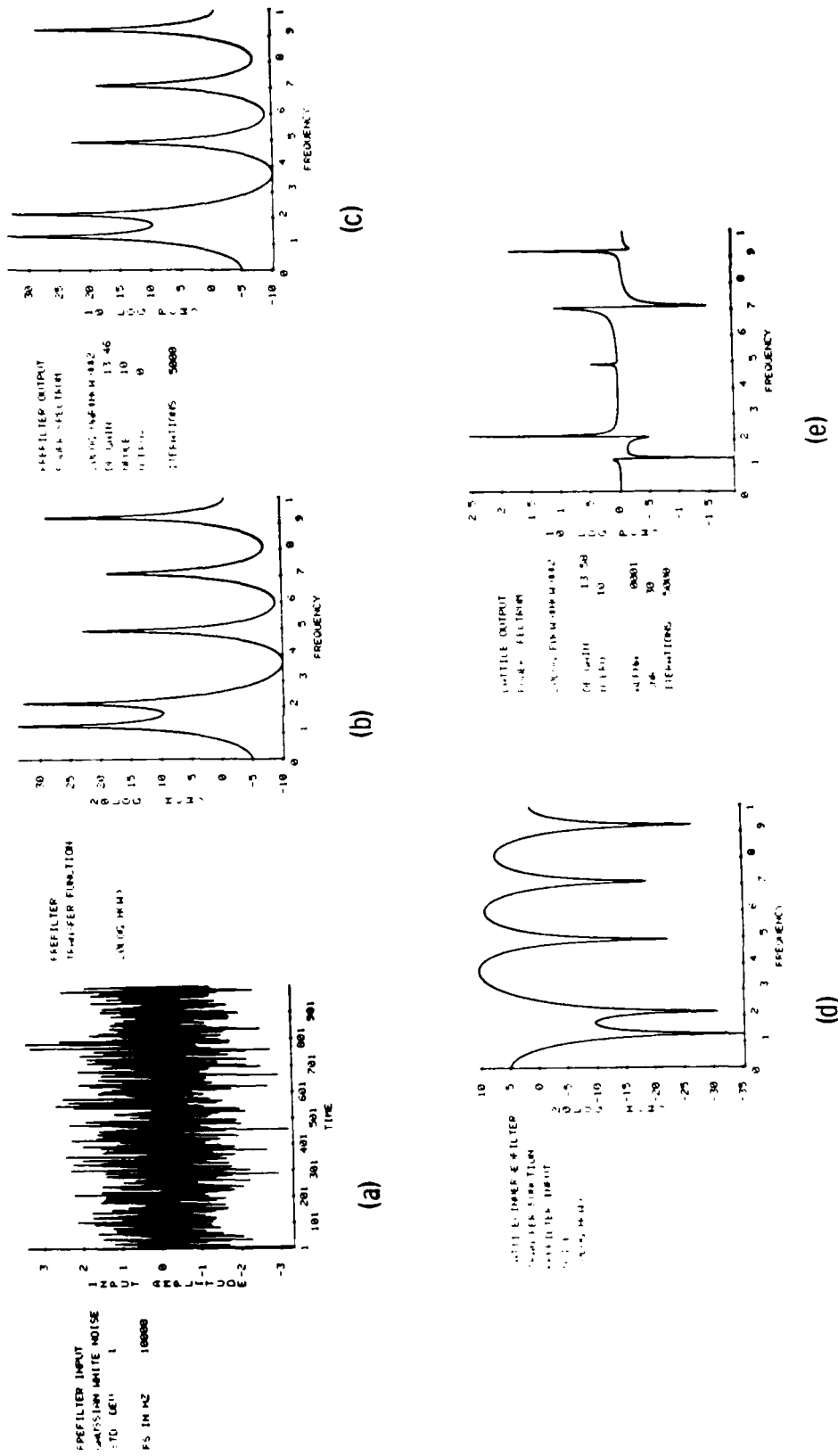


Fig. 6. Each state of the system (see Fig. 2) for a noise input to the five-formant case.

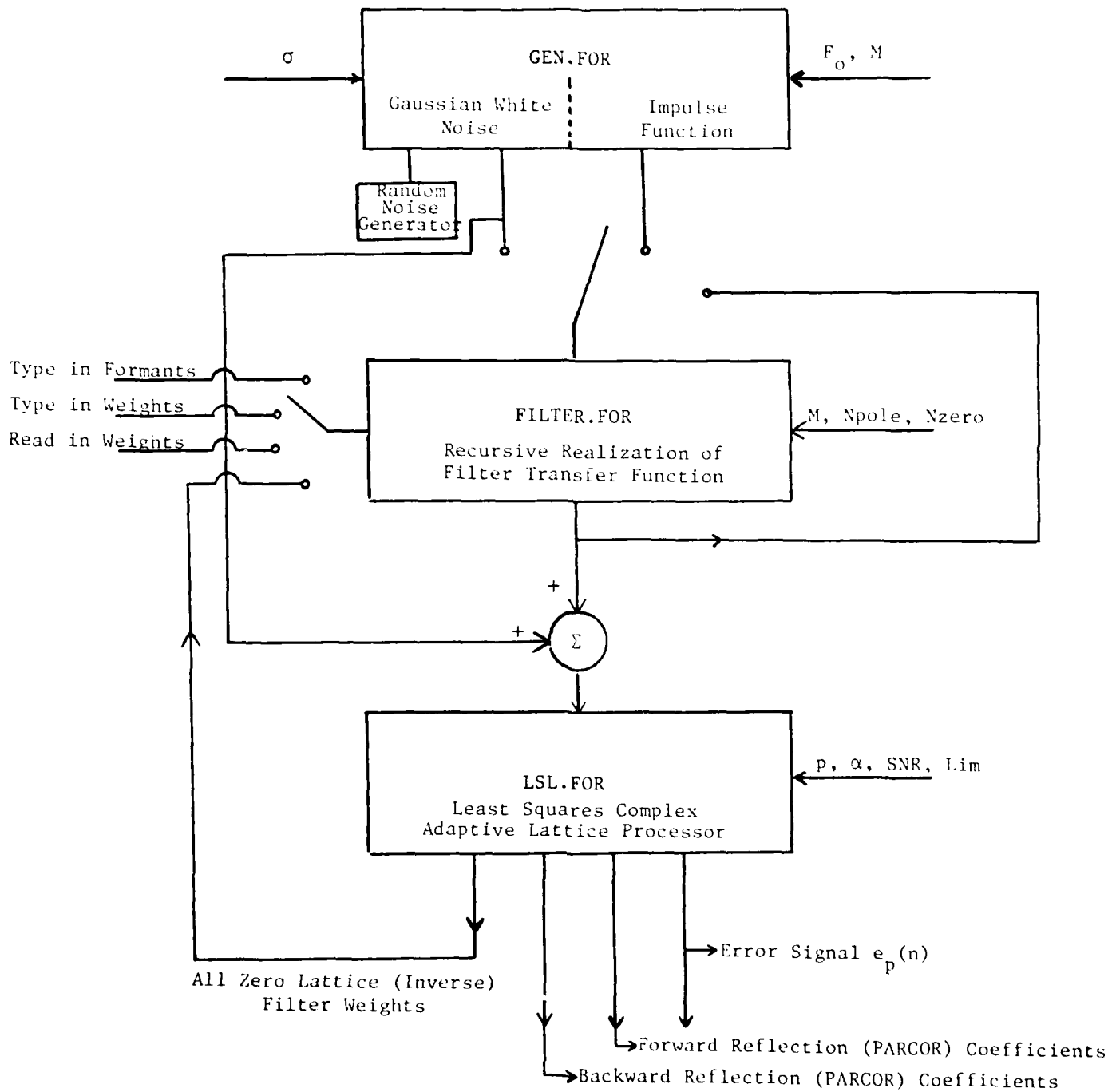


FIGURE 7. CALLING HIERARCHY AND STRUCTURE OF COMMAND PROCEDURE

TABLE I
RELATIONSHIPS BETWEEN LATTICE PARAMETERS, REF. 1 SYMBOLS, AND FORTRAN VARIABLES

Lattice (Inverse) Filter Parameter	REF. 1 SYMBOL	FORTRAN VARIABLE
Vector of forward predictor coefficients at stage i	$a_k^{(i)}(n)$ $1 \leq k \leq i-1$	A(M) $1 \leq M \leq J-1$
Highest (ith) forward predictor at state i	$a_i^{(i)}(n)$	A(J)
Vector of backward predictor coefficients at stage i	$b_k^{(i)}(n)$ $1 \leq k \leq i-1$	B(M,J) $1 \leq M \leq J-1$
Vector of backward predictors at previous stage, previous time	$b_{k-1}^{(i-1)}(n-1)$	B1(M-1,J-1)
Initialize backward predictor at stage i	$b_o^{(i)}(n)$	B(2,J)
Zero out vector of backward predictors for lower stages at time = -1	$b_k^{(i)}(-1)$ $0 \leq k \leq i-1$, $i \neq p$	B1(I,J)
Forward power at stage i	$E_i^e(n)$	EE(J)
Initialize forward power at time = n	$E_o^e(n)$	EE(2)
Backward power at stage i	$E_i^r(n)$	ER(J)
Initialize backward power for lower stages at time = -1	$E_i^r(-1)$ $i \neq p$	ER1(I)
Initialize backward power at time = n	$E_o^r(n)$	ER(2)
Forward reflection coefficient at stage i	$K_i^o(n)$	KE(J)
Backward reflection coefficient at stage i	$K_i^r(n)$	KR(J)
Forward error at stage i	$e_i(n)$	SE(J)
Initialize forward error at time = n	$e_o(n)$	SE(2)
Backward error at stage i	$r_i(n)$	SR(J)
Initialize backward error at time = n	$r_o(n)$	SR(2)
Initialize backward error for lower stages at time = -1	$r_i(-1)$ $i \neq p$	SRI(I)

TABLE 1 (Continued)
RELATIONSHIPS BETWEEN LATTICE PARAMETERS, REF. 1 SYMBOLS, AND FORTRAN VARIABLES

Lattice (Inverse) Filter Parameter	REF. 1 SYMBOL	FORTRAN VARIABLE
Stepsize parameter at stage i	$\gamma_i(n)$	DEL(J)
Initialize stepsize for each stage i at time = -1	$\gamma_i(-1) \quad i \neq 0$	DEL(1)
Input time series	$x(n)$	X(1)
Gain related parameter (gain = $(1 - \gamma_{i-2}(n-1))^{-1}$)	$\gamma_{i-2}(n-1)$	GAM(I-2)
Initialize gamma for time = n-1	$\gamma_{-1}(n-1)$	GAM(1)
Stage variable, current stage	i	J $2 \leq J \leq P+2$
Stage variable, lower stages	k	M $3 \leq M \leq J-1$
Time variable	(n)	Implicit
Time = -1, initial time	(-1)	Denoted by 1(1)
Previous time	(n-1)	Denoted by 1()
Related to fade factor, $(1 - \alpha_{CLS})$	α_{CLS}	ALSL
Filter order	P	P(in ISL), NZERO(in filter)
Number of time samples (iterations)	N	LIN(in ISL), M(in filter)

Case Figure No.	Prefilter Formant, Hz	Prefilter Bandwidth, Hz	Prefilter A_k , $k=1, \dots, N_{\text{pole}}$	Lattice Filter A_k , $k=1, \dots, N_{\text{zero}}$
3	2500.00	150.00	0.000000 0.985112	0.000000 0.995760
4	2500.00	150.00	0.000000 0.985112	-0.002642 0.985765
5	650.30 1075.70 2463.10 3558.30 4631.30	94.10 91.40 107.40 198.70 89.80	-0.266366 -0.522137 0.174152 0.247624 0.513600 0.260513 0.141718 -0.496971 -0.241914 0.943518	-0.232250 -0.543238 0.145636 0.211085 0.515390 0.260130 0.067095 -0.420906 -0.221051 0.860902
6	650.30 1075.70 2463.10 3558.30 4631.30	94.10 91.40 107.40 198.70 89.80	-0.266366 -0.522137 0.174152 0.247624 0.513600 0.260513 0.141718 -0.496971 -0.241914 0.943518	-0.263710 -0.529765 0.172208 0.254260 0.508698 0.264767 0.146636 -0.505169 -0.236822 0.941123

TABLE II

FORMANT FREQUENCIES, THEIR BANDWIDTHS, AND CORRESPONDING
FILTER COEFFICIENTS FOR THE FOUR INPUT CASES DEPICTED IN FIGS. 3-6

Command Procedure LATTICE.COM
Controls Sequence of Program Execution for VT100 Terminals

[illegible]

..... TO STOP SCULLEY, DOWN THE STREET, TO STAY AGAIN, PUSH THE SCULLEY,.....

LEARN CHARACTER, LEAST SQUARES, ADAPTIVE ALGORITHMS, PHONEME AND EDITING PACKAGE

0474 A. C111.0222

THIS PROGRAM WAS BEING OFFERED AS OF 12/6/42. THE LISTINGS SUBMITTED BY THE BUREAU OF INVESTIGATION ON 12/12/42 ARE NO LONGER CURRENT. THE LISTINGS FOR THE FOLLOWING PROGRAMS: FLETCHER, GUN, FOR, GUN, AND CIVILIAN.

3. CONSIDER CONCURRENCE AND RULES FOR COMPLETING, TAKING AND EXECUTION OF EACH PROCESS. THE CONCURRENT EXECUTION WILL BE NECESSARY FOR EACH PROGRAM TO BE RUN.

YOU MAY EDIT SOME THE PROBLEMS AT ANY TIME BY TYPING RU AFTER THE PROMPT TO RECONSTRUCT. YOU WILL BE ABLE TO OBTAIN HARD COPIES OF DATA FILES WITHIN THE RECONSTRUCTING PERIOD. BUT AFTER YOU EXIT FROM THE PROGRAM, ALL YOUR DATA FILES WILL BE DELETED OR OVERWRITTEN. BE SURE TO MAKE SOME BACKUP FILES WHICH YOU CAN RECOVER IF A DISASTER STRIKES.

TO EIGHTY. BUT HE HAD A FIRST PRIORITY LIKE TO DIE.

ERATED YOU ENJOYED MOVING AT A PLOT ON THE SCREEN, PRESS THE SPACE BAR TO
ERASE THE PLOT TO ERASE AND THE NEXT PLOT TO APPEAR.

WHEN USING THE TENDONALLY FINISHED, YOU WILL NEED TO DEPRESS THE PATTERN BUTTON THREE TIMES TO SET THE STOPS TO SPACE.

PLEASE WAIT FOR THE COMPUTER TO COMPLETE AND LEAVE THE PROGRAMS. YOU WILL RECEIVE A BEEP WHEN THE COMPUTER IS READY.

TYPE LATHE. 1.11
SPECTRA COPY SENT
11 AM COPY SENT
SPECTRA COPY
11 AM COPY
SPECTRA COPY

[illegible][illegible]

[illegible]

A COMPANY PROCESSES CONTROLS THE CRYPTING, LYING AND EXECUTION OF EACH MESSAGE. THE COMPANY RECEIVES THE MESSAGE AND CRYPTS TO BE SENT.

אשר לא ידעו, שיש להם חובות, ויש להם חובות, ויש להם חובות.

PLEASE USE THE FOLLOWING NUMBER, YOU WILL NEED TO EXPRESS THE OTHER ACTION
TO GET THE NEWS IN PLACE.

[illegible]

PAGE 01 "HIGH DODGEM THE HIGH"
 ASSIGNED TO YOU SCHEDULE: SYSDATE:
 000 100
 TABLE CHECK "HIGH VLSI TO HIGH"
 IF "HIGH" THEN, COPY TABLE
 COPY 11
 LABEL:
 TABLE CHECK "HIGH VLSI TO HIGH" COPY
 IF "HIGH" THEN, COPY LABEL

C THIS MESSAGE WAS BEEN RECEIVED AS OF 17/04/82, THE LISTINGS SUPPLIED
C WITH THE TELETYPE TRANSMISSION R2-219, DATED 11/17/82 ARE NO LONGER
C CORRECT. PLEASE OBTAIN NEW LISTINGS.

THIS PROGRAM GENERATES COMPLEX GAUSSIAN NOISE WITH VARIABLE STANDARD DEVIATION AND CORRELATION COEFFICIENT SPECIFIED BY VARIABLE FREQUENCY.

(UUCS) 25 May 1964 4130400

TYPED, RELATED NUMBER OF SAMPLES TO GENERATE.
TYPED, REUSE SAME SAMPLE CORRESPOND TO DESIGNED NUMBER OF
TREATMENTS OR FACTORS.
TYPE, N(10000=5000).
APPROX.

BYE, "LETTER TESTED STANDARD DEVIATION OF GAUSSIAN WHITE NOISE."
THEY, "LETTER TESTED STANDARD DEVIATION OF GAUSSIAN WHITE NOISE."

[illegible]

CLIPF(06916,600) "FILE
NUMBER(S),WHICHIS INFORMATION IS STORED IN DISK FILE: ,15)

THE NEXT FOUR MONTHS TAKING ALONG ON SHIRT OF ZERO.
ON 100 1=1,"

$$\begin{aligned} A_1(T) &= w_1 \mu_1 c_1 A_1 c_1^{n-1}(T) \\ A_2(T) &= w_2 \mu_2 c_2 A_2 c_2^{n-1}(T) \\ A_3(T) &= w_3 \mu_3 c_3 A_3 c_3^{n-1}(T) \\ A_4(T) &= w_4 \mu_4 c_4 A_4 c_4^{n-1}(T) \\ A_5(T) &= w_5 \mu_5 c_5 A_5 c_5^{n-1}(T) \\ A_6(T) &= w_6 \mu_6 c_6 A_6 c_6^{n-1}(T) \\ A_7(T) &= w_7 \mu_7 c_7 A_7 c_7^{n-1}(T) \\ A_8(T) &= w_8 \mu_8 c_8 A_8 c_8^{n-1}(T) \\ A_9(T) &= w_9 \mu_9 c_9 A_9 c_9^{n-1}(T) \\ A_{10}(T) &= w_{10} \mu_{10} c_{10} A_{10} c_{10}^{n-1}(T) \end{aligned}$$

21.15.15

THIS INFORMATION VARIES FOR TWO HUNDRED POINTS AT ONE TIME.
CO TO 111, 110, 1

$$\begin{aligned} \chi_1 &= \chi_1(2, 0, 1) = 1 \\ \chi_2 &= \chi_2(2, 0, 1) \\ \chi_1 &= \chi_1(2, 0, 1) \\ \chi_2 &= \chi_2(2, 0, 1) \\ \chi_1 &= \chi_1(2, 0, 1) = 5 \\ \chi_2 &= \chi_2(2, 0, 1) = 5 \\ \chi_1 &= \chi_1(2, 0, 1) = 5 \\ \chi_2 &= \chi_2(2, 0, 1) = 5 \end{aligned}$$
[illegible]

THE CALIFORNIA POWER TO DISC.

$\begin{pmatrix} 0 & 1 \\ 1 & 0 \end{pmatrix}$

THE FIRST PART OF THE PROGRAM GENERATES IMPULSES FOR CONSUMERS.

• 434011 3119 110111 011101 •

LET ARDAY TO THREE.

[illegible]

Recursive Realization of Prefilter and Lattice (Inverse) Filter Transfer Functions

C (ROW (CONTENT) IS AN ARRAY OF THE REAL PARTS OF THE FILTER COEFFICIENTS.
C (COL (CONTENT) IS AN ARRAY OF THE IMAGINARY PARTS OF THE FILTER COEFFICIENTS.

[illegible]

[illegible]

[illegible]

Program LSL.FOR Least Squares Complex Lattice Processor

THIS PROGRAM WAS DESIGNED AS OF 12/8/82. THE LISTINGS SUPPLIED WITH
ALL INFERRAT OPERATIONS 82-210, DATED 11/12/82 ARE NO LONGER CORRECT.
PLEASE OBTAIN A NEW LISTING.

COMPLEX SINGLE CHANNEL LEAST SQUARES LATTICE ALGORITHM

PROPERTY PACKAGE AND USED INTERPACK BY W.A. COMPTON.
LSL ALGORITHM BY A. COMPTON (1981).

COMMON /LSL/ F(120), F0(120), F01(20), GAM(120), AKP(16)
1, VLS(100), P01(20), S0(20), S01(20), S0(20), XXX(5000)
1, M(120), K(120), P(20), P0(20), COM(16), M(20,20), P1(20,20)
1, COM(16), S0(1000), S0(1000), S0(1000), S0(1000), S0(1000), S0(1000), S0(1000), S0(1000)

REAL AND, REAL(1000), P1(1)

INTEGER P

INTEGER N, N0, N1

DATA COM(16)/70014/

ASSIGN TO THE FILE NUMBER.
N0=1

READ F(1,1) F0(1)
F0(1)=F(1,1)

READ F(1,2) F0(2)

READ F(1,3) F0(3)

READ F(1,4) F0(4)

READ F(1,5) F0(5)

ASK THE USER FOR SOME PARAMETERS.

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5

IF F0(1) IS 0.0 THEN
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

IF F0(1) IS 0.0, OR, ALSO, F0(1) IS 0.0 THEN
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

IF F0(1) IS 0.0, OR, ALSO, F0(1) IS 0.0 THEN
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

ASK THE USER FOR SOME PARAMETERS.

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0

ACCEPT 9,5

END IF

END IF

END IF

END IF

END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0

TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

DO TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

TYPE 9,5 UNTIL F0(1) IS NOT 0.0

TYPE 9,5 UNTIL F0(1) IS NOT 0.0
TYPE 9,5 UNTIL F0(1) IS NOT 0.0
ACCEPT 9,5
END IF

END IF

107
107
107

100

— 28 —

10

10

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BAC-LHS:cac

SUBROUTINE SYNTH(N,M)

THIS SUBROUTINE SYNTHESIZES A COMPLEX POLYNOMIAL IN Z**N-1 FROM AN
ARRAY OF THE POOTS OF THE POLYNOMIAL.

N IS THE ORDER OF THE POLYNOMIAL.
B (OUTPUT) IS AN ARRAY OF THE FILTER COEFFICIENTS OF THE POLYNOMIAL
IN Z**N-1.

```

45      COMPLEX A(16),B(17),ANEG(16)
        PI=4.0*ATAN(1.0)
        DO 555 I=1,N
          CONTINUE
        DO 10 K=1,N
          B(K)=(0.0,0.0)
          CONTINUE
        R(2)=-1.0*A(1)
        R(1)=(1.0,0.0)
        DO 20 I=2,N
          DO 10 J=(I+1),2,-1
            R(J)=R(J)-A(I)*R(J-I)
          CONTINUE
        R(1)=A(1)
        R(2)=A(2)
        CONTINUE
        RETURN
        END

```

SUBROUTINE SYNTH(N,FF,BRW,HR,CC,FSS)

C THIS PROGRAM HAS BEEN REVISED AS OF 1/24/83. THE LISTINGS SUPPLIED
C WITH ARL INTERNAL MEMORANDUM R2-219, DATED 11/12/82 ARE NO LONGER
C CORRECT. PLEASE OBTAIN NEW LISTINGS.
C *****

C THIS SUBROUTINE SYNTHESIZES A COMPLEX POLYNOMIAL IN Z**N-1 FROM AN
C ARRAY OF FORMANT FREQUENCIES AND AN ARRAY OF THEIR RESPECTIVE BANDWIDTHS.
C N IS THE NUMBER OF FORMANTS AND THE DEGREE OF THE POLYNOMIAL.
C FF IS AN ARRAY OF THE FORMANT FREQUENCIES IN HZ.
C BRW IS AN ARRAY OF THEIR BANDWIDTHS IN HZ.
C HR (OUTPUT) IS AN ARRAY OF THE REAL PARTS OF THE FILTER COEFFICIENTS.
C CC (OUTPUT) IS AN ARRAY OF THE IMAGINARY PARTS OF THE FILTER COEFFICIENTS.

C SEE TEXT FOR RELATIONSHIPS BETWEEN FORMANTS, THEIR BANDWIDTHS,
C AND THE FILTER COEFFICIENTS.

```

      COMPLEX A(16),B(17),ANEG(16),PFXP,ANEXP,F2P,E2N
      REAL HR(17),CC(17),FF(R),BRW(R)
      PI=4.0*ATAN(1.0)
      DO 107 I=1,(N/2)
        FF(I)=FF(I)/FSS
        BRW(I)=BRW(I)/FSS
        E1=EXP(-PI*BRW(I)*PI)
        PFXP=CMPLX(0.0,(2.0*PI*FF(I)))
        ANEXP=CMPLX(0.0,(-2.0*PI*FF(I)))
        F2P=EXP(PFXP)
        E2N=EXP(ANEXP)
      C THESE ARE THE ZPLANE POLES.
      A(I*2)=F1*E2P
      A(I*2-1)=F1*E2N
      CONTINUE
      DO 30 K=1,17
        R(K)=(0.0,0.0)
      CONTINUE
      R(2)=-1.0*A(1)
      R(1)=(1.0,0.0)
      DO 20 I=2,N
        DO 10 J=(I+1),2,-1
          R(J)=R(J)-A(I)*R(J-I)
        CONTINUE
      CONTINUE
      DO 92 M=1,N
        R(M)=R(M)+A(1)*R(M+1)
        CC(M)=A(1)*R(M+1)
      CONTINUE
      RETURN
      END

```

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```

SUBROUTINE ADDRESS(ADDR,ISD)
C THIS SUBROUTINE SETS THE ADDR COUNTDOWN OF THE ELEMENT WITH THE MAXIMUM
C VALUE.
C ADDR IS IN ADDR OF BITS.
C ISD IS IN ISD OF THE ADDR.
      DIMENSION ADDR(100)
      ADDR=0
      DO 10 I=1,100
        IF (ADDR(I).GT.ADDR(I+1)) THEN
          ADDR=ADDR(I)
        END IF
      CONTINUE
      RETURN
END

```

```

SUBROUTINE ADDRESS(ADDR,ISD)
C THIS SUBROUTINE SETS THE ADDR COUNTDOWN VALUE IN THE ADDR.
C ADDR IS IN ADDR OF BITS.
C ISD IS IN ISD OF THE ADDR.
      DIMENSION ADDR(100)
      ADDR=0
      DO 10 I=1,100
        IF (ADDR(I).GT.ADDR(I+1)) THEN
          ADDR=ADDR(I)
        END IF
      CONTINUE
      RETURN
END

```

```

SUBROUTINE ADDRESS(ADDR,ISD)
C THIS SUBROUTINE SETS THE ADDR COUNTDOWN VALUE IN THE ADDR.
C ADDR IS IN ADDR OF BITS.
C ISD IS IN ISD OF THE ADDR.
      DIMENSION ADDR(100)
      ADDR=0
      DO 10 I=1,100
        IF (ADDR(I).GT.ADDR(I+1)) THEN
          ADDR=ADDR(I)
        END IF
      CONTINUE
      RETURN
END

```

```

SUBROUTINE ADDRESS(ADDR,ISD)
C THIS SUBROUTINE SETS THE ADDR COUNTDOWN VALUE IN THE ADDR.
C ADDR IS IN ADDR OF BITS.
C ISD IS IN ISD OF THE ADDR.
      DIMENSION ADDR(100)
      ADDR=0
      DO 10 I=1,100
        IF (ADDR(I).GT.ADDR(I+1)) THEN
          ADDR=ADDR(I)
        END IF
      CONTINUE
      RETURN
END

```

APPENDIX B

SAMPLE TERMINAL SESSION FOR THE CASE ILLUSTRATED IN FIG. 3

```

LATTICE
WHICH PROGRAM TO USE?: GEN
$ENTER NUMBER OF SAMPLES TO GENERATE.
$THIS NUMBER SHOULD CORRESPOND TO DESIRED NUMBER OF ITERATIONS OF LATTICE.

$(C1<=NK=5000).
5000
$ENTER DESIRED STANDARD DEVIATION OF GAUSSIAN WHITE NOISE.
$(MUST BE FLOATING POINT).
1.0
$ENTER DESIRED FREQUENCY OF IMPULSE FUNCTION IN HZ.
125.0
$WHITE NOISE IN FILE 1.
$GAUSSIAN NOISE IN FILE 2.
$IMPULSE INPUT IN FILE 3.
$MAKE SURE TO RECORD THESE NUMBERS.
FORTRAN STOP
ENTER Y/YES/ N/NO TO CONTINUE: Y
WHICH PROGRAM TO USE?: FILTER
$ENTER 1 IF YOU WOULD LIKE TO READ CALCULATED INVERSE FILTER COEFFICIENTS
FROM
DISK.
$ENTER 0 IF YOU WOULD LIKE TO TYPE IN FILTER COEFFICIENTS.
$ENTER 2 IF YOU WOULD LIKE TO SPECIFY FORMANTS AND BANDWIDTHS.
$ENTER 3 IF YOU WOULD LIKE TO READ IN PREFILTER COEFFICIENTS FROM DISK.
2
$ENTER NUMBER OF SAMPLES TO ANALYZE.
$THIS NUMBER SHOULD CORRESPOND TO DESIRED NUMBER OF ITERATIONS OF THE LATTICE.
ICE (M
<=5000).
5000

```

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\$IF YOU WISH TO SPECIFY FORMANTS AND BANDWIDTHS, THERE WILL BE TWO (COMPLE
N CONJ
UGATED) POLES FOR EACH FORMANT.
\$ENTER NUMBER OF POLES (1<=NPOLE<=16).

I= 1
\$ENTER FORMANT FREQUENCY IN HZ.
2500.0
\$ENTER BANDWIDTH OF THIS FORMANT FREQUENCY IN HZ.
150.0
\$FILTER COEFFICIENTS IN ORDER OF INCREASING POWER.
\$KACO=1.0).

0.000000 0.000000
0.985112 0.000000

\$ENTER 1 IF YOU WOULD LIKE TO SAVE THIS INFORMATION ON DISK.
\$OTHERWISE, ENTER 0.

1
\$ENTER NUMBER OF DISK FILE TO SAVE THIS IN.
\$USE A FILE NUMBER BETWEEN 40 AND 60, INCLUSIVE.
40

(SCREEN WILL ERASE)

THIS INFORMATION STORED IN DISK FILE 40
 PREFILTER BANDWIDTH
 FORMANT 150.00
 2500.00
 FILTER COEFFICIENTS (INCREASING POWER, $AB=1$).
 0.000000 0.000000
 0.985112 0.000000
 #ENTER DESIRED INPUT FILE NUMBER.
 #NOISE INPUT: FILE 2
 #IMPULSE INPUT: FILE 3

OUTPUT IN FILE: 21
 #BE SURE TO RECORD THIS NUMBER.
 #ENTER 1 IF YOU WOULD LIKE TO PLOT INPUT TIME SERIES.
 #ENTER 0 OTHERWISE.
 1
 #ENTER FIRST AND LAST TIME POINTS OF INPUT TO PLOT.
 (1<=M<=) 5000
 1
 149
 #ENTER INTERVAL BETWEEN POINTS ON PLOT.
 1

(PLOT (a) APPEARS ON SCREEN)

(DEPRESS CR OR SPACE BAR ONCE OR TWICE TO ERASE PLOT)

#ENTER 1 IF YOU WOULD LIKE TO PLOT THE INPUT AGAIN ON A DIFFERENT SCALE.
 #OTHERWISE, TYPE IN 0.

0
 #ENTER 1 IF YOU WOULD LIKE TO PLOT OUTPUT TIME SERIES.
 #OTHERWISE ENTER 0.

1
 #ENTER FIRST AND LAST TIME POINTS OF OUTPUT TO PLOT.
 (1<=M<=) 5000

400
 400

#ENTER INTERVAL BETWEEN POINTS ON PLOT.

1
 (PLOT (c) APPEARS ON SCREEN)

#ENTER 1 IF YOU WOULD LIKE TO PLOT THE OUTPUT AGAIN ON A DIFFERENT SCALE.
#OTHERWISE, TYPE IN 0.
0
#ENTER 1 IF YOU WOULD LIKE TO PLOT TRANSFER FUNCTION.
#OTHERWISE ENTER 0.
1

(PLOT (b) APPEARS ON SCREEN)

FORTRAN STOP
ENTER YES TO CONTINUE: Y
WHICH PROGRAM TO USE? LSL
ENTER FILTER ORDER (1<=P<=16):
2
ENTER ALPHA(CLSL):
#0.001
SPECIFY DESIRED SNR IN DB AS A REAL NUMBER.
20.0
ENTER NUMBER OF ITERATIONS DESIRED.
#11(=5000).
5000

FORWARD PREDICTOR COEFFICIENTS IN FILE: 12
BACKWARD PREDICTOR COEFFICIENTS IN FILE: 13
FORWARD PARCOR COEFFICIENTS IN FILE: 14
BACKWARD PARCOR COEFFICIENTS IN FILE: 15
FINAL FORWARD PREDICTOR COEFFICIENTS IN FILE: 11
ERROR SIGNAL IN FILE: 16
FINAL FORWARD PARCOR COEFFICIENTS IN FILE: 17
FINAL BACKWARD PARCOR COEFFICIENTS IN FILE: 18
#MAKE SURE TO RECORD THESE NUMBERS.
#IF USING THE TEKTRONIX TERMINAL, DEPRESS PAGE BUTTON
#TO CLEAR SCREEN FOR HARD COPY. (DON'T DO THIS YET)
#ENTER NUMBER OF FILE TO SAVE COEFFICIENTS IN.
#PLEASE USE A FILE NUMBER BETWEEN 70 AND 99, INCLUSIVE.
70

(PRESS PAGE BUTTON NOW TO CLEAR TEKTRONIX SCREEN)

```

FILTER FOR PREFILTER INPUT=IMPULSE
FINAL BACKWARD PARCOR COEFFICIENTS
  0.000000  0.000000
  0.995760  0.000000
FINAL FORWARD PARCOR COEFFICIENTS
  0.000000  0.000000
  0.995458  0.000000
FORWARD PREDICTOR COEFFICIENTS:
  0.000000  0.000000
  0.995760  0.000000
2-1PLANE ZEROES (ABS. VALUE):
  1.002127
  1.002127
FORTRAN STOP
ENTER 'YES' TO CONTINUE: Y
WHICH PROGRAM TO USE?: FILTER
#ENTER 1 IF YOU WOULD LIKE TO READ CALCULATED INVERSE FILTER COEFFICIENTS
FROM DISK.
#ENTER 0 IF YOU WOULD LIKE TO TYPE IN FILTER COEFFICIENTS.
#ENTER 2 IF YOU WOULD LIKE TO SPECIFY FORMANTS AND BANDWIDTHS.
#ENTER 3 IF YOU WOULD LIKE TO READ IN PREFILTER COEFFICIENTS FROM DISK.
1
#ENTER 1 IF YOU WOULD LIKE TO PLOT OUTPUT TIME SERIES.
#OTHERWISE ENTER 0.
1
#ENTER FIRST AND LAST TIME POINTS OF OUTPUT TO PLOT.
  (1<=M<=) 5000
365
435
#ENTER INTERVAL BETWEEN POINTS ON PLOT.
  1

```

(THIS PLOT NOT SHOWN IN FIG. 3; IT IS EQUIVALENT TO PLOT (e))

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#ENTER 1 IF YOU WOULD LIKE TO PLOT THE OUTPUT AGAIN ON A DIFFERENT SCALE.
#OTHERWISE, TYPE IN 0.

0

#ENTER 1 IF YOU WOULD LIKE TO PLOT ERROR SIGNAL.
#OTHERWISE ENTER 0.

1

#ENTER FIRST AND LAST TIME POINTS OF ERROR TO PLOT.
(1<=M<=) 5000

355

435

#ENTER INTERVAL BETWEEN POINTS ON PLOT.
1

(PLOT (e) APPEARS ON SCREEN)

ENTER 1 IF YOU WOULD LIKE TO PLOT ERROR AGAIN ON ANOTHER SCALE.
#OTHERWISE, TYPE IN 0.

0

#ENTER 1 IF YOU WOULD LIKE TO PLOT TRANSFER FUNCTION.
#OTHERWISE ENTER 0.

1

(PLOT (d) APPEARS ON SCREEN)

FORTRAN STOP
ENTER YES] TO CONTINUE: H
ENTER YES] FOR HARD COPY OF ALL FILTER COEFFICIENTS: H
ENTER YES] TO DELETE ALL DATA FILES: H
#

(END OF COMMAND PROCEDURE)

DISTRIBUTION LIST FOR ARL TM 82-219, by B. A. Cooper and L. H. Sibul,
dated November 12, 1982

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Department of the Navy
Washington, DC 20362

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