# COMMUNICATION SCIENCES

# AND

# ENGINEERING

# IX. SPEECH COMMUNICATION*


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## A. SPEECH AND AUDIO COMPUTER-AIDED EXAMINATION AND ANALYSIS FACILITY

The hardware of the PDP-9 Computer Facility[1] of the Speech Communication Group of the Research Laboratory of Electronics can be operated with appropriate software in a manner that provides a facility for the examination and measurement of features of acoustical waves. As an example of some of the possibilities we include in this report an abridgement of Speech Computer Facility Memo No. 7, which describes a command program called DWAS (standing for Display Waves And Spectra). This is intended to exemplify techniques that may be of value to those who wish to write their own special-purpose programs, and as a standard facility for those who can use it as a readily available instrument without concerning themselves with details of computer programming. For the second class of users it should be easier to use than a sound spectrograph.

The sound spectrograph has been the instrument most often used for acoustical measurements in speech research for the past two decades, and thus a comparison of the DWAS facility with the spectrograph may be in order. For measuring numerical values of most time- and frequency-domain phenomena the DWAS facility is faster and more accurate. One can quickly scan an analog (audio) tape looking for or at specific cases without having to tediously make a spectrogram for each 2 sec worth of data. Also the simultaneous presentation of both a time waveform and its associated spectrum enables one to correlate features in the time domain with features in the frequency domain. Also, some features such as the timing of voice onset relative to stop release are much more evident in the time domain than in a spectrogram. Because of its form of representation, however, a spectrogram gives most people a better global or over-all view of 2 sec of a sound segment. Thus, depending upon the acoustical features of interest, either


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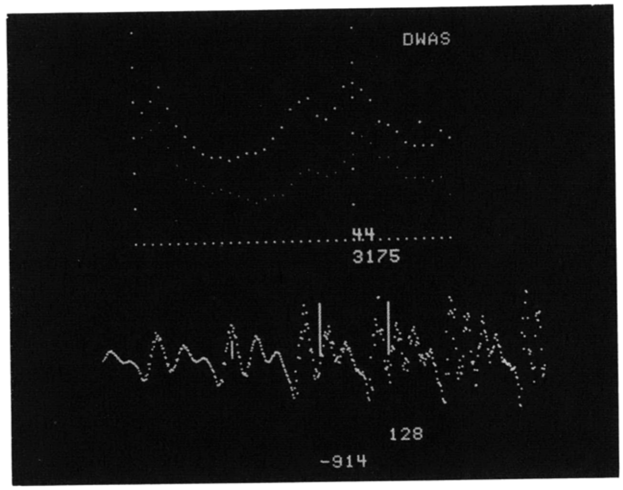


Fig. IX-1. Typical display of DWAS showing a segment of a sound waveform at the bottom and two spectra at the top.

the sound spectrograph of DWAS of both may be the best analysis facility.

Figure IX-1 shows a typical DWAS display, and the various readouts are discussed below.

DWAS is also a demonstration of the real-time multiprocessing capacities of a typical medium-sized general-purpose digital computer, since sound output in real time, the running of a spectrum analyzer, the maintenance of a display on a CRT, and the hard copy output of data on a teletypewriter can all occur simultaneously.

The abridgement of the memo now follows. Complete copies are available from the Speech Communication Group.

ABSTRACT

DWAS — Display Waves and Spectra — is a facility for the examination and measurement of both time- and frequency-domain characteristics of signals in the audible frequency range. DWAS inputs and outputs sampled waves as real-time audio, displays the sampled waves on a CRT, takes spectral samples using a hardware filter-bank spectrum analyzer and displays them, and provides various display and hard copy digital readouts of time and spectral data.

These data can be used to determine acoustical features such as pitch, relative timing and/or duration of events as visually determined from either waveform or spectral features (e.g., timing of voicing onset relative to stop release), and characteristics of formants and their transitions.

The normal display of DWAS is of a split-screen format with the bottom half showing the sampled waveform and various time-domain markers, and the top half pertaining to spectral representations of signals, i.e., outputs of the standard filter-bank spectrum analyzer.

## Time Domain Display

An analog wave is represented internally digitally, i. e. , as a sampled and digitized signal. Input signals are digitized by a linear analog-to-digital convertor (ADC), and output analog signals are generated from the digitized representation by using a digital-to-analog convertor (DAC). Input and output analog signals should normally be lowpass filtered at one-half the sampling rate to prevent aliasing (sampling theorem). (DWAS can be used to demonstrate aliasing very graphically, both aurally and visually, by using a spectral display.)

Near the center of the waveform display window is a vertical line used as a time marker. Below this line, which we call the spectral time mark, is displayed an integer that is the time in milliseconds from an arbitrary time origin. The effective position of this mark is adjustable by moving the waveform display window (see below), and thus durations and relative timings of events observable in the waveform can be quickly determined. A knob adjustable time mark somewhere to the right of the center time mark can be used for pitch measurements (see below).

In addition to repetitive playback (audio output from a digitized signal) of the entire signal there are playback modes that either terminate or initiate playback at the spectral time mark. These modes are useful for aurally locating desired segments of the signal.

All time-domain digital readouts are based upon an assumed sampling rate that must be the same as the actual sampling rate for the readouts to be valid. The actual sampling rate is selected by the rotary switch "CLOCK2," and the assumed rate can be reset via teletype upon button request.

## Frequency-Domain Display

Spectra can be obtained from the digitized signal by converting it to analog form and feeding that to the hardware filter-bank spectrum analyzer (filter-bank characteristics are listed in Speech Computer Facility Memo No. 1). For a single short-term spectrum the output of the filter bank is sampled and digitized at the time indicated by the spectral (center)time marker in the waveform display. Since the filter-bank outputs are smoothed by 10-ms lowpass filters, the time "window" for the spectrum resembles that of an exponential with a 10-ms time constant rising from the left (past time) and terminating at the spectral time mark. To remind the user of this spectral time window a short vertical time mark is displayed 10 ms ahead of, i. e. , to the left of, the spectral time mark.

Every time a spectral sample is taken the data update the main spectrum display. Also in the spectral display is another spectral sample called the auxiliary spectrum which is displayed with less brightness. The auxiliary spectrum is updated from the main spectrum by button command, and is useful for visually comparing different spectra.

Notice that frequency components above one half of the sampling rate are spurious, and so for a sampling rate of 10 kHz the output of filters with center frequencies above 5 kHz is of little value. For a sampling rate of 20 kHz the full range of the spectrum analyzer up to 7 kHz is valid, under the assumption that the analog signal is lowpass filtered accordingly.

## DWAS Operation

### Push Button Commands

35 RETURN – Escape from DWAS back up one level in the command hierarchy.

33 RECORD ON/OFF – Sample, digitize, and store an input signal (from MPX4). The effect is that of a tape loop in that current input records over earlier input so that the most recent input is preserved when RECORD is turned off. Alternate button pushes start and stop this mode.

32 RESET TIME ORIGIN so that zero time is redefined to be the time of the current spectral time marker.

30 SPECTRUM – Take a spectral sample of the waveform at the spectral time mark and display it as the main spectrum.

15 SAMPLING RATE – read in from the teletype a new assumed sampling rate. (Type in integer followed by a carriage return.) Clock switch "CLOCK2" which determines the actual sampling rate should also be set to this rate.

18 NUMERICAL PRINT-OUT of current main spectrum.

19 GRAPHICAL PRINT-OUT of current main spectrum.

28 PLAY ALL – Repetitively play out as real-time audio the whole waveform.

27 PLAY HEAD – Repetitively play out the waveform from its beginning up to the center (spectral) time mark.

29 PLAY TAIL – Repetitively play out the waveform from the center time mark to its end.

34 PLAY OFF – Turn off real-time audio output of the signal.

31 AUXILIARY SPECTRUM UPDATE – Copy the main spectrum into the auxiliary spectrum which is displayed with lower brightness. This is useful for comparing spectra.

21 SPECTRA – Begin repetitively taking and displaying spectra of the waveform at the spectral time mark.

22 EARLIER SPECTRUM – Move the wave display window so as to examine the waveform 10 ms earlier, and then take a spectrum.

23 LATER SPECTRUM – Move the display window so as to examine the waveform 10 ms later, and then take a spectrum.

### Knob Functions

1 – Position of time waveform display window.

2 – Spectral cursor adjustment. At the bottom of the cursor is displayed the value of the filter output in dB and the center frequency of the filter in Hz.

3 – Pitch marker adjustment. The inverse of the time separation between the adjustable pitch time marker and the spectrum time marker (fixed at center of display) is calculated and displayed as a pitch value (assuming the standard sampling rate). This is used to make pitch measurements by adjusting under visual control the position of the pitch time marker to be 1 period after the spectrum time marker.

W. L. Henke

### References

1. W. L. Henke, "Speech Computer Facility," Quarterly Progress Report No. 90, Research Laboratory of Electronics, M. I. T., July 15, 1968, pp. 217-219.

## B. TASS – ANOTHER TERMINAL ANALOG SPEECH SYNTHESIS SYSTEM

A parametric (terminal analog) speech synthesis system has been written to run on the recently acquired PDP-9 Computer Facility of the Speech Communication Group of the Research Laboratory of Electronics.[1] The synthesizer signal circuit is realized as a digital filter-based simulation rather than in analog hardware, as was a previous terminal analog synthesizer.[2] The present digital simulation cannot run in real time,

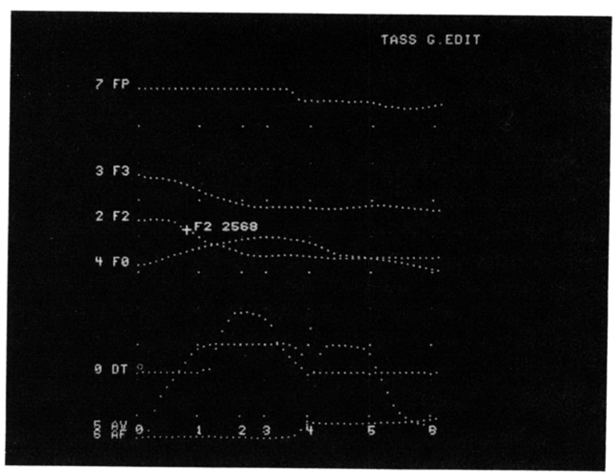


Fig. IX-2. Typical display of the TASS graphical editor during usage. The stylus is used to update parameter track "F2" as seen from its echo in the display. The value of that particular parameter for that specific ordinate value is also displayed. Note also the nonconstant value of the epoch duration parameter "DT", and how this determines the time scale as indicated by the abscissa values.

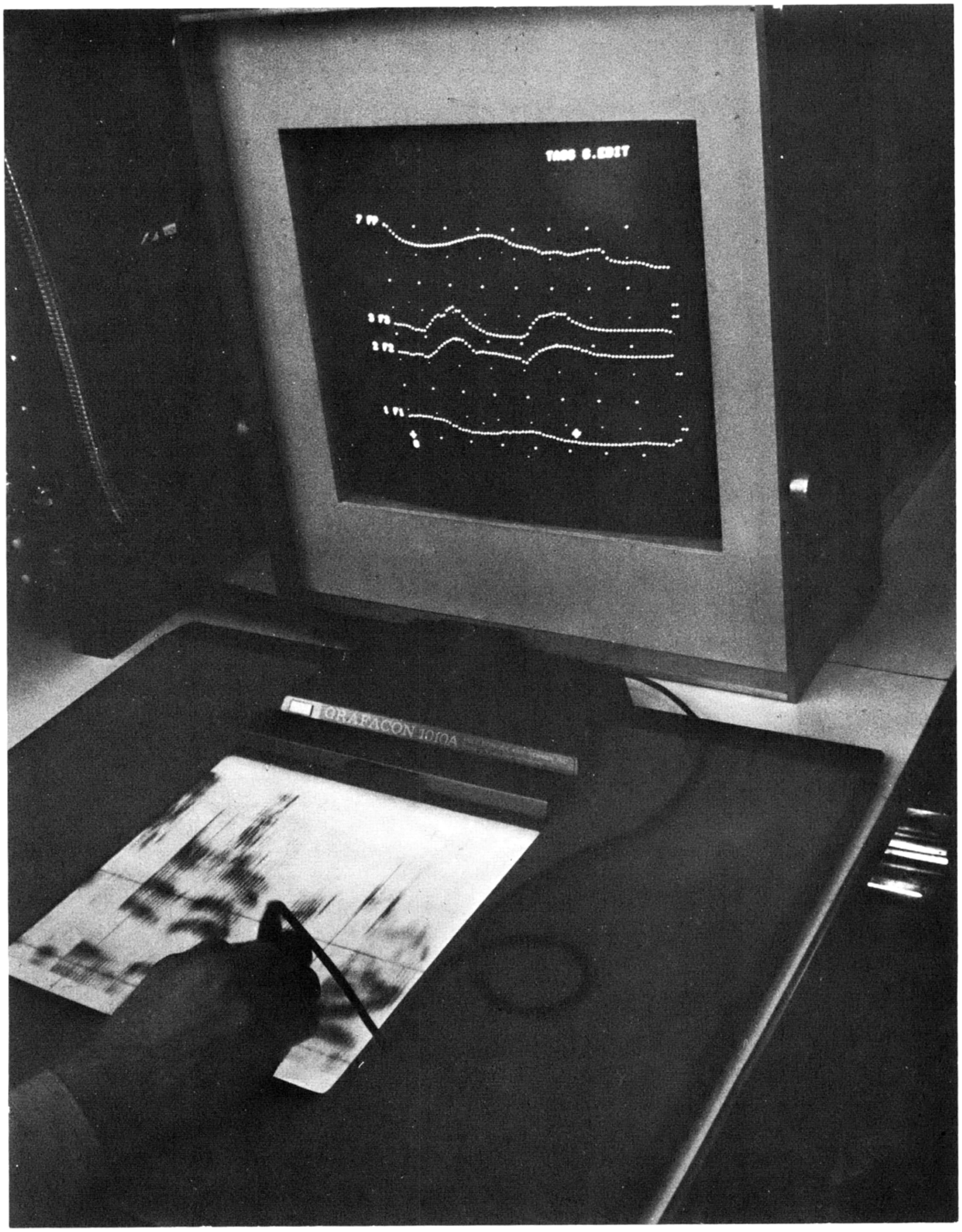


Fig. IX-3. TASS graphical editor in use. A speech spectrogram is being projected upon the bottom surface of the tablet input device by using an opaque projector and a mirror. A software calibration procedure is provided wherein the image and the display can be calibrated to each other via a transformation of stylus position.

being a factor of 5-10 times slower, and may have a slightly inferior signal-to-noise ratio than a well-tuned analog hardware synthesizer would, but because it is realized as a computer program it offers the benefits of flexibility and ease of maintenance.

Two slightly unusual features of the synthesizer parameter editor are the availability of an epoch duration parameter (seen in Fig. IX-2) and spectrogram projection facilities (Fig. IX-3). This parameter is very useful for changing only the temporal patterns of synthesized utterances, and as such has already been used extensively in the generation of a family of synthetic stimuli for an experiment on the perception of stress.

We include in this report a Speech Computer Facility memo that describes the system from a user's viewpoint. Not discussed are some of the finer details of the digital realization of the signal circuit.

## ABSTRACT

The TASS system is a facility for the parametric (terminal analog) synthesis of speech. The nominal or standard TASS system configuration currently consists in three distinct phases: (i) a graphical editor for terminal analog parameter tables, (ii) a parametric type of speech synthesizer realized digitally in software as a sampled data system, and (iii) a wave-analysis phase that outputs and inputs sampled waves as real-time audio, visually displays the sampled waves, and takes spectral samples with the spectrum analyzer and displays them.

A description of the system and instructions for its use are given. In addition to its primary function as an educational and research tool in speech science, TASS is also a good case study in techniques of effectively utilizing the available hardware of the Speech Computer Facility, with emphasis on graphical, acoustical, and digital (buttons, knobs, switches, teletypewriters, etc.) man-machine interaction.

## PARAMETRIC (TERMINAL ANALOG) SPEECH SYNTHESIS

"Terminal Analog" is the generic name for types of speech synthesizers which accept as control (input) data time-varying acoustical parameters of the desired signal. The family of parameters is dependent upon the configuration of the synthesizer (usually best represented in block-diagram notation), which in turn is dependent upon the designer's goals and/or theories of speech production. Almost all such synthesizers will include among the control parameters formant frequencies, fundamental frequency, and voicing amplitude. Other parameters will often pertain to acoustical descriptions of speech features, such as frication, aspiration, and nasalization.

## TASS SYSTEM

A group of processes that are useful for parametric speech synthesis has been developed and integrated together into what we call the TASS (for Terminal Analog Speech Synthesis) system. The nominal or standard TASS system configuration, at present, consists in three distinct phases: (i) a graphical editor for terminal analog parameter tables, (ii) a parametric type of speech synthesizer realized digitally in software as a sampled data system, and (iii) a wave-analysis phase that outputs and inputs sampled waves as real-time audio, visually displays the sampled waves, and takes spectral samples with the spectrum analyzer and displays them.

### Tass System Operation

The graphical editor is at the highest (master) control level, and other phases can be reached from it. Upon initiation of TASS (i.e., loading the program into the computer) the user finds himself talking to the editor. The SYNTHESIZE push button will cause a transfer to the synthesizer phase which will synthesize an output waveform from the current parameter table and then transfer control to the output phase when finished.

Various hardware inputs (e.g., push buttons) have different interpretations when talking to different phases. Interphase transfer is usually invoked by push-button command. Button 35 has a conventional meaning in all phases which is to return (i.e., escape) from the present phase to the invoking phase, i.e., pop up one stage in the phase hierarchy, and it is ignored if already at the master level.

## GRAPHICAL EDITOR FOR PARAMETER TABLES

Parameter tables contain data specifying the values of individual parameters as functions of time. The data are in matrix form with the number of rows equal to the number of controlled parameters and each column specifying parameter values for a short epoch of time. There is one additional parameter which is the length of the epoch to be associated with each column. Such a quasi-parameter is useful for changing the temporal pattern of a synthesized utterance without altering its frequency and amplitude characteristics. The graphical editor provides for the input and rapid modification of all parameter tracks, including that of epoch duration.

The tracks of any parameters may be selectively and simultaneously displayed on a CRT, and modified by using the stylus of the Grafacon. Knobs are used to control the position and magnification of the editing "window" so that both relatively macroscopic and microscopic features can easily be observed and edited. Speech spectrogram images can be projected directly upon the Grafacon surface by using an opaque projector, and the editor includes functions that effectively spatially "calibrate" the system to scales on the spectrogram, i.e., frequency and time, and thus allow direct tracing of spectrograms. Such a function could also be used for rapid "computerizing" of spectrographic data for other types of analyses.

The parameter tables may be stored in the standard file system, and may be kept on any form of bulk storage – usually DECtape with the current Speech Computer system configuration. Thus in addition to being accessible to the editor and synthesizer, the tables are also available to other programs for generation (e.g., synthesis by rule) or analysis (e.g., formant track studies).

### Tass Graphical Editor Operation

The editor displays parameter tracks with time as the abscissa. A time grid is overlaid and the displayed abscissa values are times (in decisecond units) from the beginning of the parameter table. TOGA switches select the indices (row numbers) of the parameter tracks to be displayed. Attached to the first point of each displayed track is its row index and a mnemonic for its function. (Functions are attributes of specific synthesizer configurations, and are described therewith.)

The modification of any parameter track requires the conjunction of (i) track in display, (ii) track index number push-button depressed, and (iii) Grafacon stylus depressed. When these conditions are satisfied the numerical value of the associated parameter is shown adjacent to the stylus echo in the display, and that value will update the referenced parameter track data. Since the ranges and units of different parameters are different, there can be no single ordinate scale and this method of positioning the stylus and depressing the desired push button can be used to determine the value of any datum.

### Push Button Commands

0-17 UPDATE – Used to modify parameter tracks.

21 SHORTEN parameter table by deleting last columns.

22 LENGTHEN parameter table by adding trailing columns.

23 CONSTANT – Set a whole row to a constant. Row index and value used are those last referenced by an updating action.

25 SYNTHESIZE – Go to synthesis phase and calculate an output wave from current parameter table, then transfer to output phase upon completion of synthesis.

26 OUTPUT – Go to output phase directly without synthesis, thus any previous output wave is undisturbed.

27 FILE – Read and/or write parameter tables using file system, commands accepted from teleprinter (see below).

### Knob Functions

Except when in the so-called calibrated mode the knobs have the following functions:

Knob 1 – parameter table display window position.

Knob 2 – parameter table display time-scale expansion.

### Filling of Parameter Tables

Upon reception of a FILE button command the teleprinter will respond with "TAPFS?" and await a file command. The file commands are:

P – PUT out the current parameter table as a named file.

G – GET a named file and make it the current parameter table, then return to the graphical editor.

* – RETURN to the graphical editor.

When file names are needed they will be requested by a "FILNAM?" which should be answered by an up to 6-character primary name followed by an optional 3-character secondary name. All keyboard input must be terminated by a carriage return. The input/output device used for the files is currently set to tape transport number 2.

## TASS PARAMETRIC (TERMINAL ANALOG) SYNTHESIZER

The actual waveform synthesis is realized by a process that we call the TASS synthesizer, which is a digital (i. e., sampled-data) parametric (terminal analog) synthesizer implemented in software running on the general-purpose central processor. Since the software is of highly modular design, it is a very minor undertaking to reconfigure the synthesizer or make other adjustments.

The nominal configuration of the synthesizer is as represented by a block diagram (Fig. IX-4) and comprises a cascade (series) vowel network in parallel with a fricative branch. In the literature of speech communication there has been extensive discussion of parametric synthesizer configurations (e. g., parallel vs cascade formant circuits), and the user of the TASS system should keep the ease of reconfiguring well in mind so that he does not come to regard the nominal configuration as the only or necessarily best configuration.

The major filtering functions of the synthesizer are implemented by interconnections of blocks whose transfer functions are those of a single complex conjugate pole or zero pair. Such transfer functions can be realized in the analog case by a simple 3-element RLC circuit, but are realized digitally in the TASS synthesizer by a few multiplications and additions plus two unit delays.

The signal-processing network is implemented by using 18-bit integer arithmetic (single-precision on a PDP-9 computer) and demonstrates that an adequate, although not exceptional, signal-to-noise ratio can be achieved for similar configurations by using 18-bit arithmetic. The sample rate is 10 kHz. This particular configuration runs 5-10 times slower than real time on a PDP-9 computer, the synthesis rate being somewhat variable dependent upon input parameters, since unexcited circuits are ignored in the simulation.

### Synthesizer Operation

A synthesis is executed by a command to the editor (see editor commands). An "output" waveform is synthesized and system control is then passed to the DWAS phase for

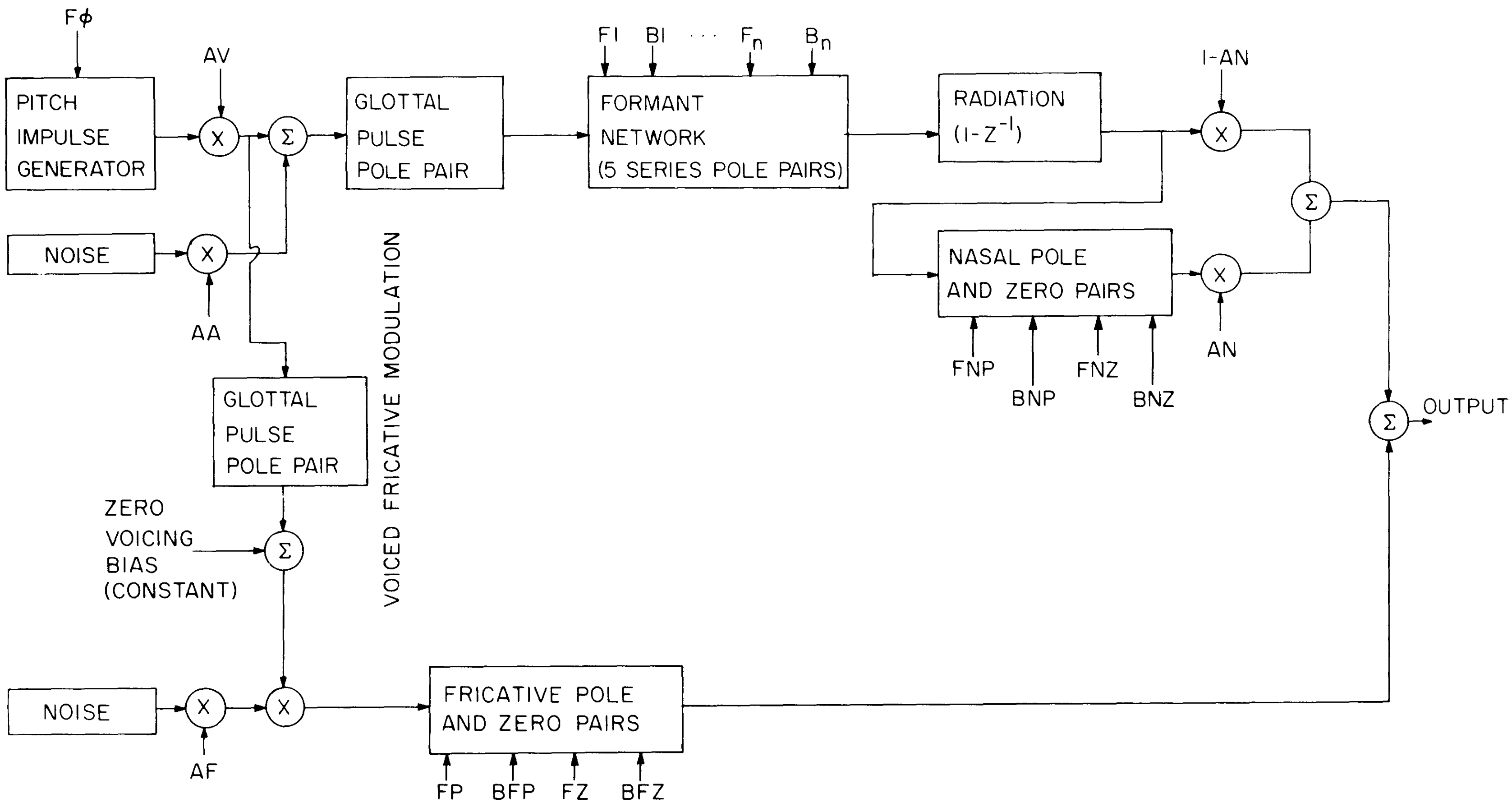


Fig. IX-4. Typical parametric speech-synthesizer configuration.

aural and visual examination of the new signal.

Certain parameter values can cause a signal overload. The most common reasons for this are tuning several resonators (formants) to almost the same frequency or tuning a zero pair to too low a frequency (since in this case the design requirement of unity gain at DC forces the gain to become excessive). When an overload occurs the synthesis is aborted and system control is returned to the editor. An error message is output which reads "OVERFLOW AT time," where time is the number of milliseconds of (synthetic) time from the beginning of the synthesis when the overload occurred. The corresponding time in the parameter table can then be examined to locate the offending value.

## Synthesizer Parameters

Those synthesizer parameters that in general will be time-varying are attached to a row in the input parameter table. Other parameters have been given an initial constant value and are not included in the table to conserve table space. For specific purposes the attachment of any synthesizer parameter to a parameter table row can be consummated by a trivial program change.

Parameter units are as follows, and particular values are displayed at the stylus location during input/edit when using the editor.

| | |
|---|---|
| frequency | Hz |
| time | msec |
| amplitude | 0 to 1 linear, displayed as 0 to 131,071 |

CONTROLLED PARAMETERS

| mnemonic | index (row number) | function |
|---|---|---|
| DT | 0 | delta time, duration for this column |
| F1 | 1 | first-formant frequency |
| F2 | 2 | second-formant frequency |
| F3 | 3 | third-formant frequency |
| F0 | 4 | fundamental frequency (glottal pitch) |
| B1 | 8 | first-formant bandwidth |
| AV | 5 | amplitude of voicing |
| AF | 6 | amplitude of frication |
| AA | 14 | amplitude of aspiration |
| FP | 7 | fricative pole frequency |
| BFP | 15 | bandwidth of fricative pole |
| FZ | 16 | fricative zero frequency |
| BFZ | 17 | bandwidth of fricative zero |
| AN | 9 | amplitude of nasalization |

| | | |
|---|---|---|
| FNP | 10 | frequency of nasal pole |
| BNP | 11 | bandwidth of nasal pole |
| FNZ | 12 | frequency of nasal zero |
| BNZ | 13 | bandwidth of nasal zero |

PRESET CONSTANT PARAMETERS

| mnemonic | value | function |
|---|---|---|
| B2 | 70 | second-formant bandwidth |
| B3 | 100 | third-formant bandwidth |
| F4 | 3500 | fourth-formant frequency |
| B4 | 175 | fourth-formant bandwidth |
| F5 | 4500 | fifth-formant frequency |
| B5 | 281 | fifth-formant bandwidth |
| | 200 | glottal source shape pole frequency |
| | 250 | glottal source shape pole bandwidth |

### TASS OUTPUT PHASE

For a description of the usage of this phase see the description for the program DWAS (Display Waves And Spectra). That includes a patch panel wiring diagram. When used with TASS button 35 causes an escape from DWAS back to the TASS graphical editor.

Notice that synthetic spectra can be simultaneously compared visually with real spectra by first recording a real signal and taking spectra and then synthesizing a signal and looking at its spectra.

W. L. Henke

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1. W. L. Henke, "Speech Computer Facility," Quarterly Progress Report No. 90, Research Laboratory of Electronics, M. I. T., July 15, 1968, pp. 217-219.
2. R. S. Tomlinson, "SPASS – An Improved Terminal Analog Speech Synthesizer," Quarterly Progress Report No. 80, Research Laboratory of Electronics, M. I. T., January 15, 1966, pp. 198-205.

## C. A FACILITY FOR STUDYING PERCEPTION OF TIMING IN NATURAL SPEECH

Some earlier experiments on the perception of timing in natural speech[1-3] showed that, in some contexts, changing the duration of a phoneme in a spoken sentence had no effect on the perceived duration of an adjacent phoneme, whereas in other contexts an equal and opposite change had to be made for the timing of the sentence to sound natural. A factor that appeared to be important in deciding which result obtained was the position of the experimentally manipulated phonemes relative to the stressed syllables in the sentence. The stimuli for the experiments mentioned above were produced by an electronic equivalent of tape cutting and splicing. The sentence was recorded on two tracks of a loop of tape, and played back through two separate heads. A second recording was made from the first, and, at the desired place in the sentence, the output of the first recorder was switched from one playback head to the other, so that a segment of the signal was repeated or omitted. This method can be applied only to segments that have a steady-state spectrum, such as can occur in unvoiced stops and fricatives and in some voiced sounds when the context is very carefully controlled. This was an irritating shortcoming, and we also desired to expand the earlier studies.

Therefore, a general-purpose program for editing audio signals has been written for our group's PDP-9 computer. About 3 sec of speech can be stored in a buffer in the core of the computer, and the buffer can be subdivided into segments by means of 18 markers that can be placed on the buffer. The markers and the stored waveform can be displayed on an oscilloscope, and the time axis of the display can be expanded so that a marker can be placed on any desired even-numbered sample (i.e., with a maximum error of 0.0001 sec). The placing of markers can also be checked by ear, since any segment or sequence of segments of the waveform defined by the markers can be reconstituted by a D-to-A converter driving an audio transducer. Each segment can be played between zero and three times before proceeding to the next, by setting the appropriate switches in three banks of toggles, or the number of plays can be set to any larger number by means of the AC switches. This makes it possible to alter the duration of a speech segment whose spectrum is changing: A marker is placed at the start of each glottal cycle in the segment, and selected glottal cycles can be repeated or omitted so that the desired effect is obtained.

Edited waveforms can be put out and recorded on analog tape, or they can be stored in digital form on DECtape. On DECtape, the waveform in the buffer (or any part of it) is stored separately from the markers that define the segments and the number of times that each segment is to be played. This minimizes the storage used when several editions are made from one master waveform.

In addition to the above functions, any defined segment of the buffer can be cleared, or filled from DECtape or any signal source that is connected to the input A-to-D converter. The sequence of samples in the buffer, or any segment of it, can be cyclically permuted, which permits re-ordering of the segments, among other things. The spectrum of the waveform can be taken at any desired point, and two such spectra can be displayed simultaneously, for comparison. A number also appears on the display, representing the duration in fifths of a millisecond between any selected marker and the cursor at the midline of the oscilloscope display.

The only shortcoming of the present system is the size of the buffer – which, however, is large enough to hold a sentence of reasonable length. The storage capacity of the computer is to be enlarged soon, and it may then be possible to combine the waveform-editing program with a software parametric synthesizer (see Sec. IX-B), so that segments with exactly controlled properties could be synthesized to replace segments of natural speech.

I would like to acknowledge my indebtedness to W. L. Henke. Many of the features of the present program were made possible by the design of the Speech Communication Group's computer facility,[4] and its feasibility was suggested by an earlier, more elementary, tape-splicing program.[5]

A. W. F. Huggins

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1. A. W. F. Huggins, "How Accurately Must a Speaker Time His Articulations?," IEEE Trans., Vol. AU-16, pp. 112-117, 1968.

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5. W. L. Henke, "AWAVE – Audio Wave Displayer and Splicer," Memo No. 3, Speech Communication Group, Research Laboratory of Electronics, M. I. T., 1968.