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Robotics Control Using Isolated Word Recognition of Voice Input

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16. Abstract

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The speech output facility incorporates a commercially available voice synthesizer based upon phonetic representations of words. The same DEC PDP-11/03 microcomputer used in the voice input system controls the speech output operation.

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PREFACE

The work described in this report was performed in partial fulfillment of the Master of Science in Computer Science Degree at the University of California, Los Angeles, under the cognizance of the Earth and Space Sciences Division of the Jet Propulsion Laboratory.

ACKNOWLEDGMENTS

I wish to express my gratitude to Dr. B. Bussell, Dr. A. Klinger and Dr. P. Ladefoged of my committee for their encouragement and guidance throughout the course of this work. I also thank Dr. Bussell, my thesis advisor, for his willingness in my undertaking this particular project.

I also extend my appreciation to my colleagues in the Robotics Research Group at the Jet Propulsion Laboratory, where most of this work was undertaken. In particular, I would like to thank Dr. D. Williams for the great deal of time he spent in discussing problem areas with me and for his valuable suggestions. I thank Ray Eskenazi for his efforts in designing and constructing the speech preprocessing hardware.

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ABSTRACT OF THE THESIS

A speech input/output system is presented that can be used to communicate with a task oriented system. Human speech commands and synthesized voice output extend conventional information exchange capabilities between man and machine by utilizing audio input and output channels.

The speech input facility described is comprised of a hardware feature extractor and a microprocessor implemented isolated word or phrase recognition system. The recognizer offers a medium sized (100 commands), syntactically constrained vocabulary and exhibits close to real-time performance. The major portion of the recognition processing required is accomplished through software, minimizing the complexity of the hardware feature extractor.

The speech output facility incorporates a commercially available voice synthesizer based upon phonetic representations of words. The same DEC PDP-11/03 microcomputer used in the voice input system controls the speech output operation.

CHAPTER 1 - USE OF AUDIO FOR ROBOTICS CONTROL

Generally, man-machine communication is in a form consistent with the operational requirements of the machine rather than in a form convenient to the user. Keyboard input and hard copy output are examples of such interactions that can be replaced by audio communication. Advantages inherent in voice control arise from its universality and speed. Speech exhibits a high data rate for an output channel. The human voice is also the best form of interactive communication when an immediate reaction is desired. Voice input and output help provide a flexible system of communication between the computer and user. Speech permits the hands, eyes and feet to remain free, allows the operator to be mobile and can be used in parallel with other information channels.

The idea of automatic recognition of speech is not new. At the time of this research limited word recognition systems have been used in industry; some implemented systems have also incorporated voice output to provide two-way audio man-machine communication. Trans World Airlines, Inc. and United Air Lines, Inc. use speech input in some of their baggage sorting facilities [HERS 73].

Voice input systems are also used by shippers to separate and route parcels [GLEN 71, NIPP 76], in numerically controlled machine tool programming to specify part descriptions [MART 76], and in compressor repair facilities to record serial numbers of air conditioning components returned for service. Some air traffic controllers and aircraft crew members are trained on simulators which incorporate speech input and synthesized voice output [GLEN 75, GRAD 75]. Automatic word recognizers and speech output devices enable the telephone to be used in a conversational manner to query, access, and modify remote data base systems [BEET 00]. Voice recognition techniques have been applied in security systems to recognize or verify the identities of persons on the basis of their speech patterns [ATAL 72, BEEK 71, BEEK 00]. Other examples of speech output devices include automatic text readers for the visually handicapped and the audio reporting of credit or account information for retail stores and banks [DATA 74]. Simple speech recognition systems are currently available which can handle a vocalulary of 15-150 words and cost from $10,000 to $20,000 [GLEN 75].

The work presented in this report is directed at the design and implementation of a voice input/output facility to be used to communicate with the robotic systems at the Jet Propulsion Laboratory, Pasadena, California. The robot

system (figure 1.1) is a breadboard, intended to provide a tool for testing various approaches to problem-solving and autonomous operation [LEWI 77]. The major components of the integrated system include perception(vision), path planning, locomotion, manipulation, simulation and control. The processors which perform these operations (figure 1.2) include a remote Decsystem 10, a General Automation SPC-16/85 minicomputer, an IMLAC PDS-1D graphics display system and three DEC PDP-11/03 microcomputers. One PDP-11/03 with a floppy disk drive serves as the microcomputer network coordinator. The second PDP-11/03 is used as a communications controller for the distributed system, and the third is used for the speech input/output interface. The voice input system is composed of both hardware and software processing which make up the isolated word recognizer. Voice output is accomplished through use of a VOTRAX VS-6.4 Audio Response System under control of the third microcomputer. This processor configuration was chosen to allow flexibility in the robotics research program.

The speech input/output system presented can be used to control the execution of a task oriented system. The application presented in this work is directed at providing a user with the capability to question, direct and simulate the performance of the JPL robot system and its individual

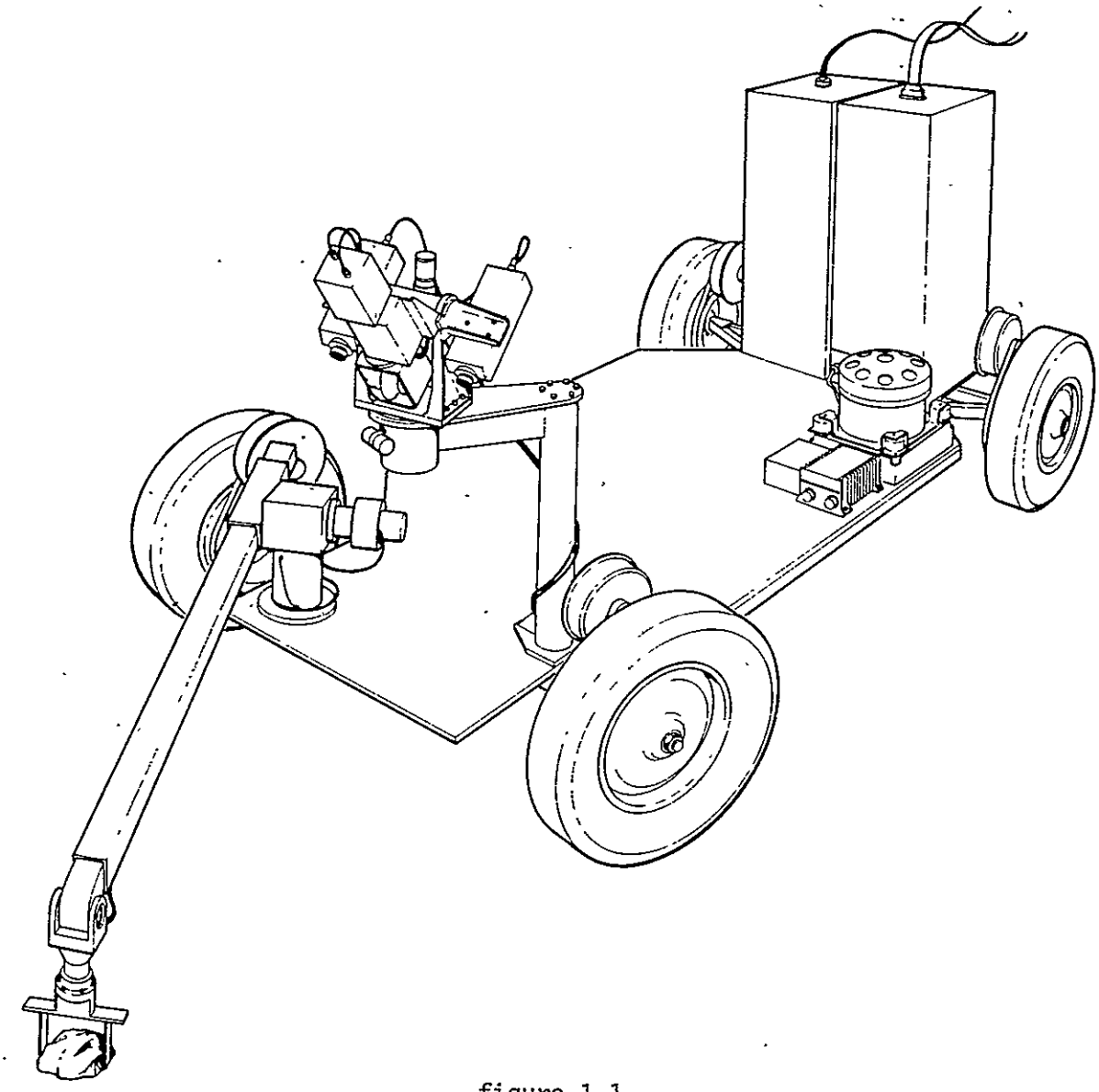
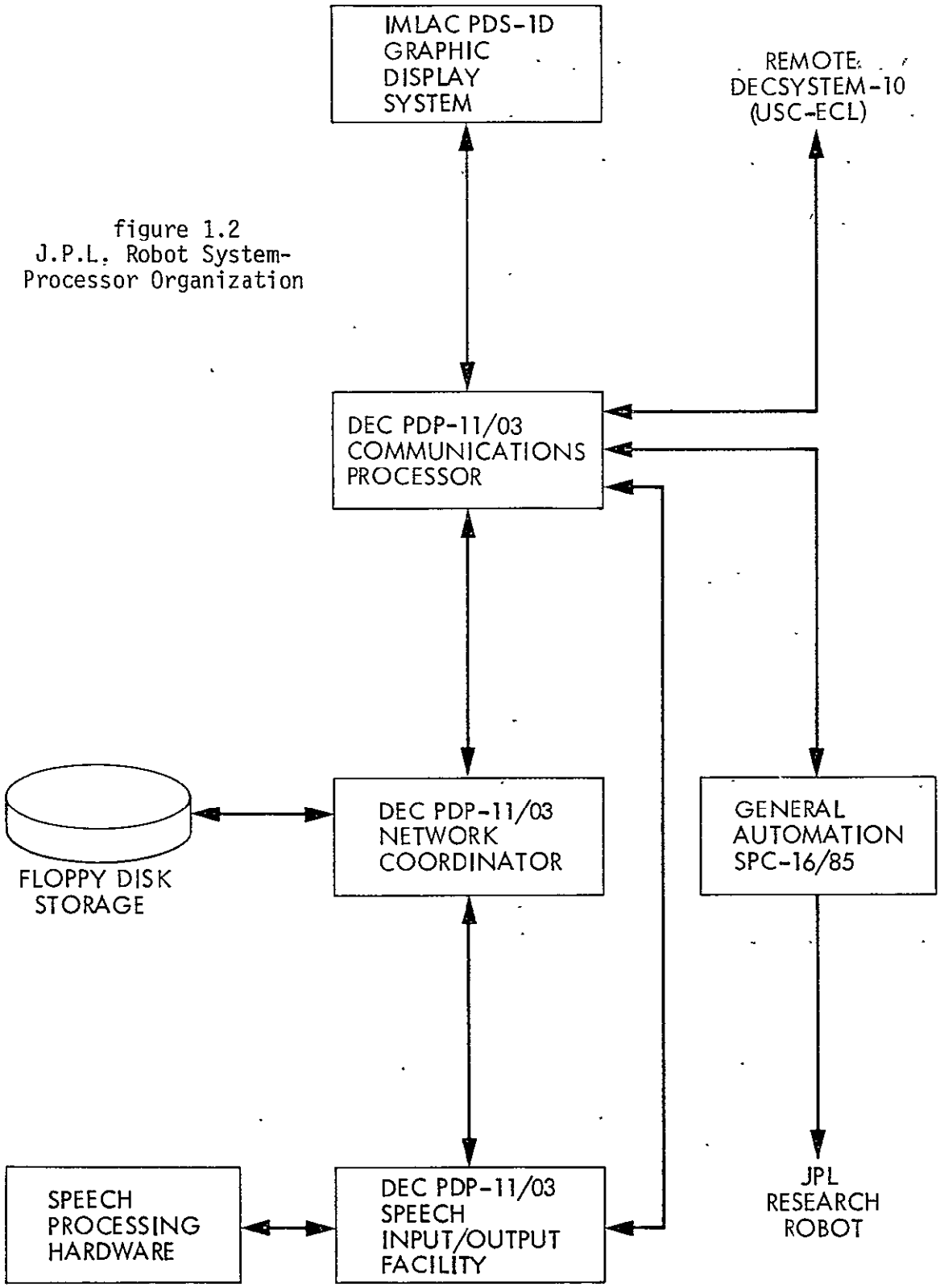


figure 1.1
J.P.L. Research Robot

figure 1.2
J.P.L. Robot System-
Processor Organization



subsystems. The IMLAC graphics system is used to display status information, predicted positions of vehicle components and terrain maps of the environment. The user, through voice input, will be able to specify the execution of local graphics transformations upon the CRT image or select a new area of interest for which a display can be created. For each subsystem status display, the user can query the data base for its specific state of activity. For example, information may be requested regarding the relative positions of obstacles lying within the planned path of the vehicle, or the user may call up an additional display routine of the arm to evaluate the performance of a set of wrist joint positions upon the grasping of an irregularly shaped object. When viewing a representation of the robot vehicle and its surroundings, the user may desire to simulate planned actions (e.g. vehicle movement, arm motion) before their actual execution. Critical system states are automatically communicated to the user through voice output. This type of man-machine interaction readily lends itself to the application of voice communication.

This report begins with a brief presentation in chapter 2 of the mechanisms involved in human speech generation and recognition. The bulk of the research however, is directed at the work involved in the development of the speech input facility and is addressed in chapter 3. The voice output

system is presented in chapter 4.

CHAPTER 2 - HUMAN MECHANISMS FOR SPEECH GENERATION AND RECOGNITION

Before beginning a design of the automatic speech recognition system, it was helpful to first gain an understanding of the mechanisms involved in human speech production and recognition. These mechanisms are qualitatively presented with attention given to their effects upon sound wave properties.

2.1 Human Speech Production

Man generates sound by causing air molecules to collide. Air is drawn into the lungs and expelled through the trachea into the throat cavity by means of the respiratory muscles. Near the top of the trachea resides two lips of ligament and muscle, the vocal cords. Voiced sounds are produced by the flow of air forcing oscillation of the vocal cords. The mass of the cords, their tension and the air pressure upon them determine the frequency of vibration.

Other components of the human speech facility which affect the accoustic properties of the generated sounds include the vocal tract, nasal cavity and mouth. The vocal tract proper is a deformable tube of non-uniform

cross-sectional area whose configuration influences the frequencies comprising the speech waveform. The movements of the lips, tongue and jaw change the size of the opening from which the air passes; this affects the nature of the signal produced, as does the person's rate of speaking, emotional state and the context of the utterance [GLEN 75].

Human speech is actually continuous in nature. The properties of the speech wave reflect the time dependent changes in the vocal apparatus. Despite this characteristic, words can be represented as strings of discrete linguistic elements called phonemes. For example, the word "boiling" is described phonetically (in [ELOV 76]) by the VOTRAX [VOTR ØØ] string, "/B//Ol//AY//I3//L//I//NG/." Standard American English contains 38 distinct phonemes [ATMA 76]. Phonemes can be divided into the categories: pure vowels, semi-vowels, diphthongs, fricatives, nasals, plosives and laterals.

Pure vowels are normally produced by a constant vocal cord excitation of the vocal tract. The tract and mouth configuration is relatively stable during the voicing of the sound. The sound is mostly radiated through the mouth; some radiation of the vocal tract walls also occurs. (The mouth is not as stable in the production of semi-vowels, such as /w/ and /y/). Diphthongs are transitions from one pure vowel to another. Fricatives, (e.g. /v/ in "vote,"

/z/ in "zoo," /h/ in "he") are produced from noise excitation of the vocal tract, such as the air flow that results when the tongue is placed behind the teeth. Nasals, (e.g. /m/ in "me," /n/ in "no") result from vocal cord excitation coupled with closure at the front of the vocal tract by the lips or tongue. Plosives result from explosive bursts of air, (e.g. /p/ in "pack," /k/ in "keep," /t/ in tent). The /l/ sound is an example of a lateral.

2.2 Human Speech Recognition

The ear is conventionally divided into three acousto-mechanical components: the outer ear, the middle ear and the inner ear. The outer ear is composed of the pinna (the large appendage on the side of the head commonly called the ear), the ear canal and the tympanic membrane (eardrum). The outer ear collects the rapid fluctuations in air pressure characterizing the sound wave, leads it down the ear canal and sets the tympanic membrane into vibration.

The middle ear cavity is filled with air and the three ossicular bones, the malleus, incus and stapes, (informally called the hammer, anvil and stirrup respectively). The function of the middle ear is to provide an impedance transformation from the air medium of the outer ear to the fluid medium of the inner ear. This amplification of the

pressure applied to the stapes footplate from the tympanic membrane is on the order of 15:1. Middle ear muscles (the tensor tympani and the stapedius) provide protection for the inner ear from excessive sound intensities by restricting the movement of the ossicles [LITT 65]. In adjusting the sensitivity of the ear, these muscles also provide a low-pass filter characteristic [FLAN 65].

The inner ear is composed of the liquid filled cochlea and vestibular apparatus and the auditory nerve terminations. The tympanic membrane as it vibrates, exerts pressure on the stapes footplate which is seated on the cochlea. This provides a volume displacement of the cochlear fluid proportional to the motion of the tympanic membrane. The amplitude and phase response of a given membrane point along the cochlea is similar to that of a relatively broad bandpass filter. Mechanical motion is converted into neural activity in the organ of Corti.

The ear appears to make a crude frequency analysis at an early stage in its processing. Mechanisms in the middle ear and inner ear seem to measure properties of peak amplitude, pitch and relative intensity of the component sound waves [FLAN 65, WHIT 76b]. For these reasons, a frequency domain representation of speech information appears justified and advantageous.

CHAPTER 3 - THE AUTOMATIC ISOLATED WORD RECOGNITION SYSTEM

Success has been demonstrated in the recognition of isolated words from a fixed vocabulary; accuracy rates in excess of 97 per cent have been reported for 50-200 word vocabularies, [BOBR 68, ITAK 75, MCDO 00, VICE 69]. The two areas of continuous speech recognition and speech understanding exhibit more difficult problems and are often confused with the area of isolated speech recognition. To clarify the use of these terms, the following definitions are given:

> ISOLATED SPEECH RECOGNITION- The recognition of single words in which a minimum period of silence is required between adjacent words (usually at least one tenth second) to insure that the adjacent words do not confuse the analysis of the current utterance.
>
> CONTINUOUS SPEECH RECOGNITION- The recognition of words spoken at a normal pace, without unnatural pauses between words to aid in end-point detection.
>
> SPEECH UNDERSTANDING- The recognition and understanding of words or phrases spoken in a natural manner in which semantic or pragmatic information is utilized.

3.1 General Description

In the design of the automatic speech recognizer process of this project, many decisions had to be made affecting its overall structure and performance. The decisions arrived at reflect the intended use of the system in addition to its possible evolution. The following speech recognition system properties characterize its robotics control application:

- single word (often mono-syllabic) or short phrase commands
- medium sized, extensible vocabulary (100 words)
- high accuracy desirable (99 per cent)
- close to real-time operation
- cooperative user environment
- single speaker used per session; different session may be directed by a different speaker
- must execute on a DEC PDP-11/03 microcomputer
- flexible software design and interface
- low cost

Throughout the design of the recognizer, these specifications were followed to produce the needed end-product. It should be stressed that this work was directed at the previously outlined application and not at the realization of a general purpose, speaker-independent, large vocabulary speech understanding system. The

development of this low-cost microprocessor process was attainable as a consequence of its task specific nature.

The single word recognition design constraint enabled the system to be developed as an isolated word recognizer. This decision reduced the difficulty of word boundary detection found in continuous speech and in speech understanding. This choice also resulted in an easier attainment of a high accuracy rate in near real-time.

The medium sized vocabulary property made necessary the development of data compression and pattern comparison operations that would permit the DEC PDP-11 microprocessor to quickly access and process the speech data. As a vocabulary increases in size, the opportunity for one word to be confused with another becomes greater. Most speech recognition systems use some form of high-level linguistic or semantic analysis to achieve an adequate rate of recognition [HATO 74]. A tree structured vocabulary for this isolated word recognizer was developed to provide near real-time, accurate recognition. This use of syntactic constraints is discussed in section 3.4.

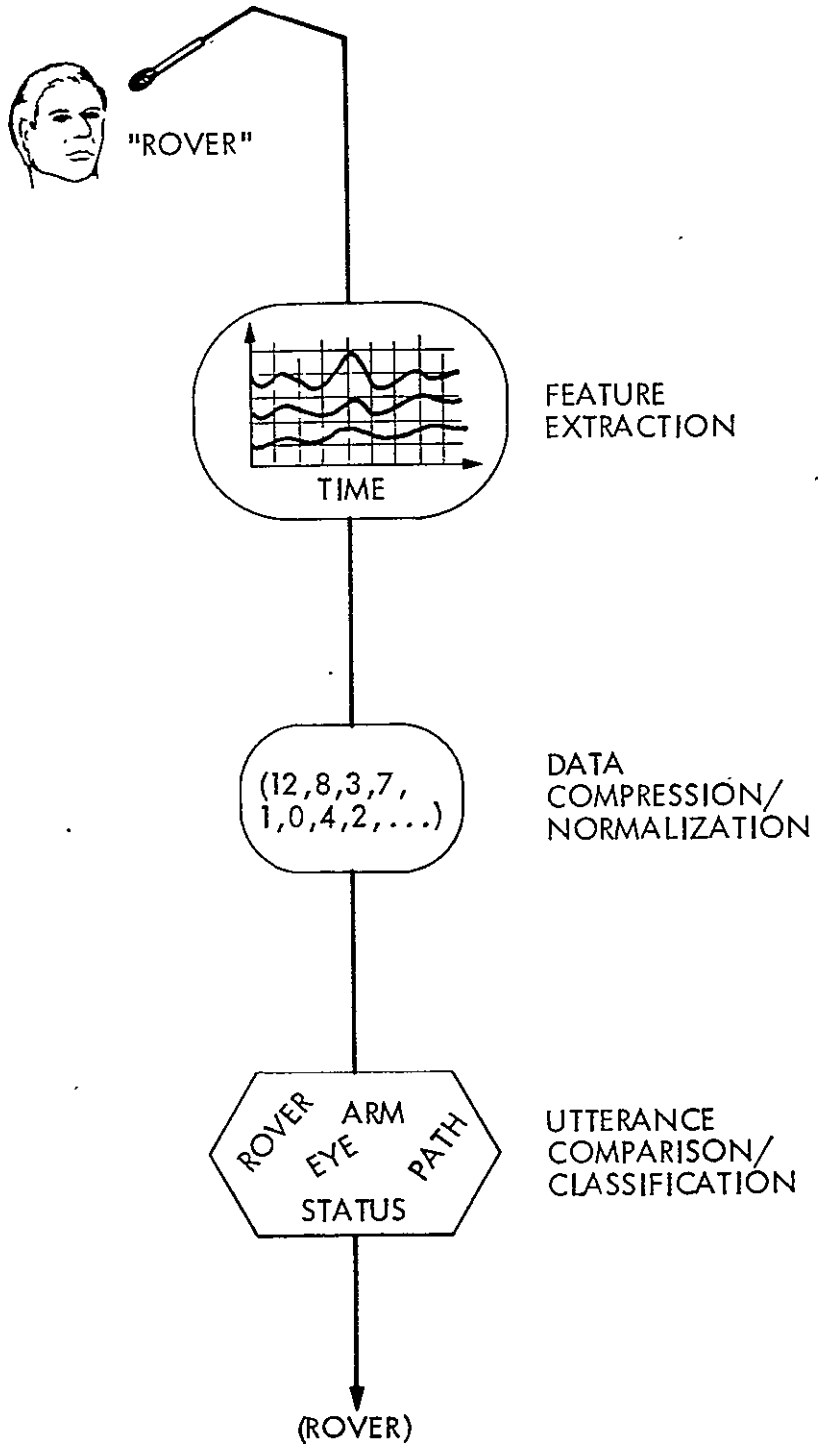
The recognition software has been written in DEC PDP-11 assembly language [DEC 76] for overall system efficiency. A flexible program architecture was realized through use of highly structured modularized routines. Firm routine

interfaces localize component responsibilities and permit individual subroutine modifications without side effects.

The isolated word recognition system can be segmented into its three main functions: feature extraction, data compression/normalization and utterance comparison/classification (figure 3.1.1). During feature extraction, the input voice signal is sampled and its representative properties measured. This results in the collection of large amounts of speech data. To permit conservation of storage and processing speed, the incoming data is compressed and normalized. Utterance matching techniques are used to identify the input pattern sample. The condensed input is compared to the stored parameterization of each word in the vocabulary. A decision is made based upon the results of the comparisons.

The choices made for the feature extraction, data compression/normalization and utterance comparison/classification procedures for the J.P.L. word recognition system were based upon system characteristics such as processor speed and instruction set, as well as vocabulary type and structure. The three sets of recogniton routines selected were required to be compatible with each other.

figure 3.1.1
Isolated Word Recognition System Components



3.2 Feature Extraction

Some form of preprocessing is required to represent the speech signal in a reasonable manner. If the signal amplitude was sampled and digitized every 50 microseconds, and one byte was required per value, 20,000 bytes of memory would be needed to record an utterance one second in duration. Real-time processing of this amount of data would be difficult, and word prototypes would consume too much storage to be kept in fast memory.

Most existing word recognition systems use one of two general preprocessing techniques: bandpass filtering or linear predictive coding. Bandpass filtering segments the speech wave in 2 to 36 (usually non-overlapping) frequency bands; it is often accomplished through hardware. When the frequency segmentation corresponds to the fundamental frequencies found in human speech production, it is called formant analysis. The outputs of these bandpass filters are then examined through hardware or software means over a given time interval. Examples of such properties measured are: zero-crossings, average amplitude, peak-to-peak amplitude, total energy, average energy and power.

A discrete word recognition system developed at McDonnell-Douglas uses a mini-computer to process amplitude information from three frequency bands in attempting to

represent utterances as a series of phonemes [MCDO ØØ]. Neroth [NERO 72] uses hardware to generate analog signals proportional to zero-crossing rates and average energy for two frequency bands. Snell [SNEL 75] has proposed to use the same approach and algorithms in a slower, more limited recognition system targeted for implementation on a microprocessor. Vicens [VICE 69] also uses amplitude and zero-crossing measures, but upon a three bandpass filtered speech system. Lowerre [LOWE 76] uses peak-to-peak amplitude and zero-crossing values in a five band speech understanding system. Itahashi uses a different type of measure, ratios of the output powers of four bands, to determine phoneme classifications [ITAH 73]. Systems by Gold [GOLD 66], and Bobrow and Klatt [BOBR 68] use 16 and 19 filters respectively to analyze the speech spectrum.

Linear predictive coding (LPC), implemented in hardware or software similarly analyzes the frequency components of speech. More computation is required than in the bandpass filtering amplitude/zero-crossing techniques, but greater data reduction is realized. The output of a LPC routine can be in the form of LPC coefficients and predictive errors. LPC coefficients are used in generating an acoustic tube model of speech in order to identify formant peaks. The linear predictive residual is defined as the error which remains when a linear predictive filter is applied to a time

series representation of speech [WHIT 76b]. It has been used to provide an efficient means to measure the similarity of two utterances [ITAK 75]. Such values can be thought of as being similar to those provided by Fourier analysis or outputs from a programmable bank of narrow bandpass filters, (Makhoul has documented a system in which 36 filters were used [MAKH 71]).

Atal, Rabiner and Sambur [ATAL 76, RABI 76, SAMB 75] use zero-crossing rate, speech energy, autocorrelation coefficients of adjacent speech samples in addition to LPC coefficients and the energy of LPC prediction error to determine speech classifications. Dixon and Silverman [DIXO 75, SILV 74] through PL/I software executing on an I.B.M. 360/91, perform a discrete Fourier transform (DFT) in addition to their LPC calculation upon digitally recorded input. Itakura [ITAK 75] uses a minimum prediction residual rule based upon a time pattern of LPC coefficients to recognize isolated words. Makhoul [MAKH 73] performs his spectral analysis of speech by the autocorrelation method of linear prediction to minimize oversensitivity to high pitched speech components.

Other feature extraction techniques used have included calculation of pitch periods [ATAL 72, REDD 67, WELC 73], software implementation of LPC or zero-crossing measures using speech waves which have been digitized and stored on

tape [PAUL 70, WASS 75, WOLF 76], hardware phase-lock loop tracking of fundamental frequencies [HONG 76], and axis-crossing detection of frequency modulated speech waves [PETE 51].

The speed and capability of the LSI-11 microprocessor, and the development cost of hardware preprocessors constrained the choice of a feature extraction method for the J.P.L. sytem. Linear predictive coding software could have been implemented on the LSI-11 microprocessor, however, its execution would not permit the word recognition system to operate in close to real-time. LPC hardware would be very expensive to develop; no source was found which had knowledge of a LPC hardware package. A flexible feature extraction processor based upon a series of bandpass filters was selected.

Experience has shown that reliable isolated word recognition systems can be built using information derived from three frequency bands adjusted so that they approximate the first three formant ranges of human speech. The formant frequencies of speech are the frequencies characterized by strong resonances and energy peaks [RICE 76]. For the J.P.L. system, the frequency ranges chosen for incorporation into the feature extractor were the ranges 200-750, 800-2250, 2300-2900, and 3000-4000 cycles per second. These values roughly represent the first three

formants and the remaining higher frequencies of speech. Two CROWN model VFX2 dual-channel filter/crossovers [CROW 00] were purchased, providing four manually adjustable bandpass filters, and set to the above ranges.

An ELECTRO-VOICE model DS35 dynamic cardioid microphone is used due to its smooth frequency response and noise rejection characteristics to provide the input to the recognizer. Proper preamplification and dynamic range control of the voice signal is partly achieved by means of a SHURE model SE30 gated compressor/mixer [SHUR 76].

The configuration at this point in the design is illustrated in figure 3.2.1. The speech wave is input by means of the microphone, amplified and filtered into four bands, all in analog form. To process the output of the filters through software on the DEC LSI-11 processor, Shannon's theorem [SHAN 49] requires a sampling rate of 8,000 times per second for the highest frequency band, (ignoring non-ideal filter characteristics). In using the same sampling rate of 8,000 times per second for the four channels, 32,000 conversions per second are required. This dictates a software interrupt loop of no longer than 30 microseconds in duration to control the analog-to-digital converter and to store the digital representation for each individual band sample. An average assembly instruction in the DEC LSI-11 processor requires approximately 8

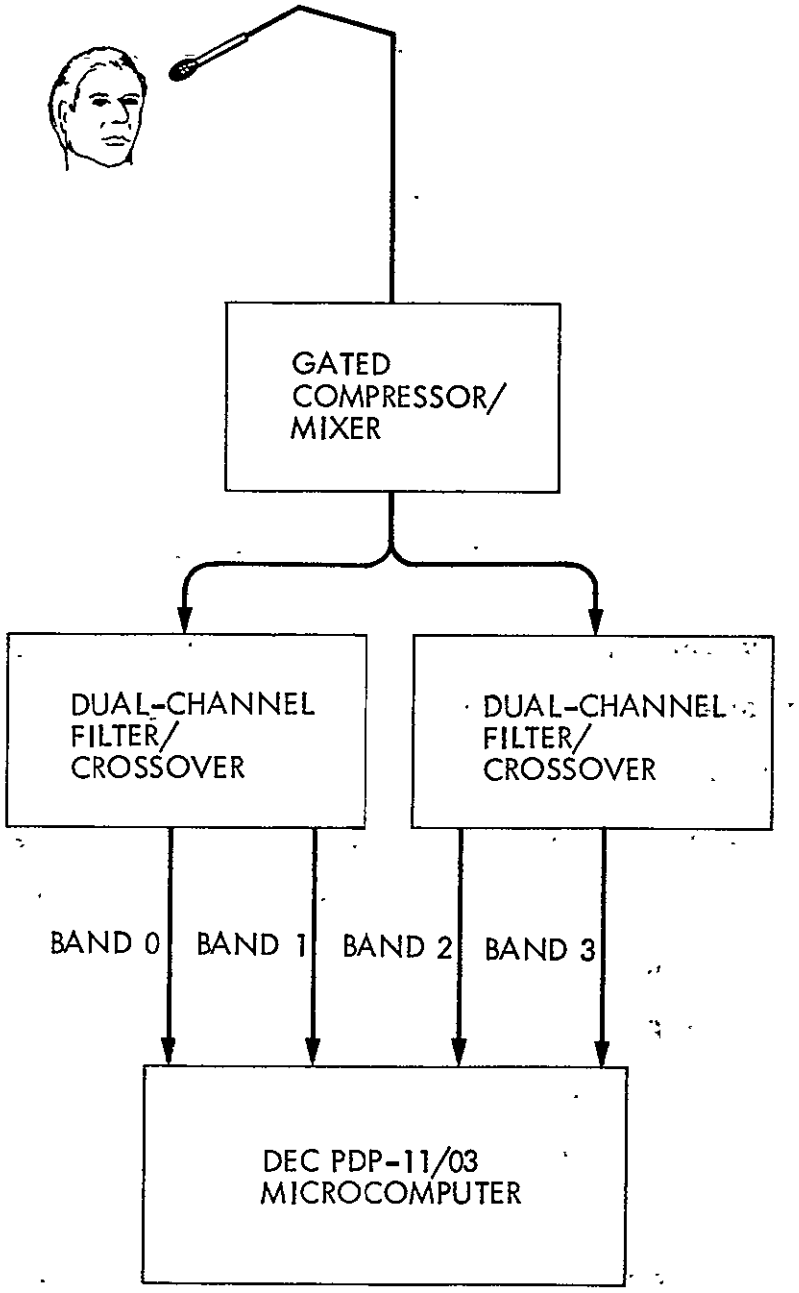


figure 3.2.1
Software Supervised Feature Extraction

microseconds to execute; this sets the maximum length of the loop at four instructions. Software data collection at these rates is impossible on the LSI-11 microprocessor. (Note: additional processor time would have been required to process the data which would fill nearly 32K words of memory per second.)

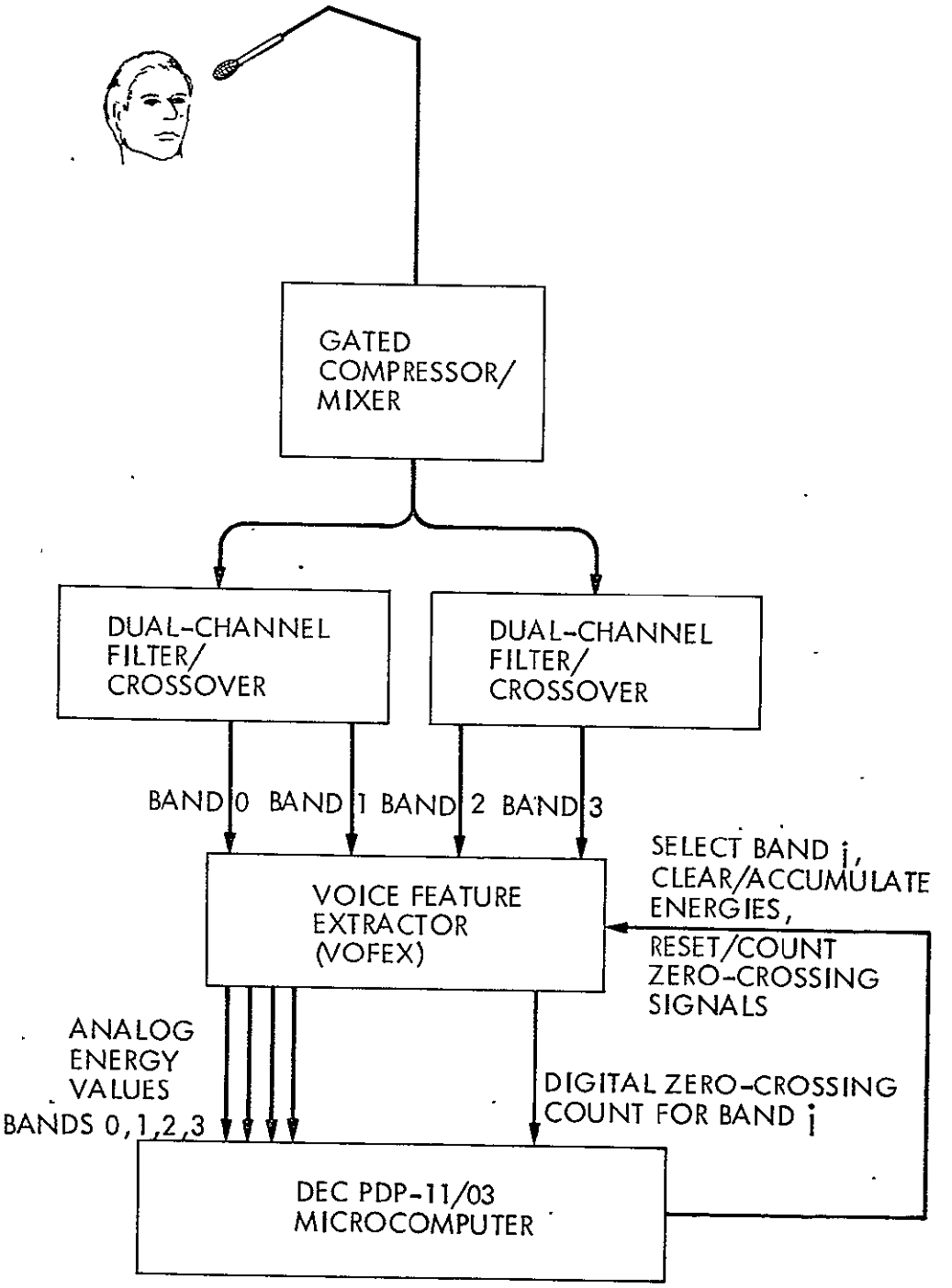
The solution to these data collection and compression problems necessitated the use of a hardware feature extractor. Of foremost importance in its design was flexibility in features measured and the ease at which it would interact with the supporting LSI-11 recognition software. A variety of measurements have been used by others in their designs of word recognition systems (and have previously been noted). Some have used zero-crossing measures together with amplitude or energy values. To allow flexibility in the choice of utterance parameterizations, the initial J.P.L. system uses a processor resettable zero-crossing counter and amplitude integrator for each of the four bands. By incorporating processor controllable counters and integrators, the software can specify the period of time for which zero-crossings will be accumulated and energy averaged. The initial "window" length chosen was ten milliseconds. Longer periods tend to provide less information regarding local properties of the speech signal (e.g. peaks) [REDD 76], while shorter periods yield

insufficient data reduction.

An easily measured property of a speech signal used by Neroth [NERO 72] in his word recognition system and by McDonell-Douglas [MCDO ØØ] in their system is the total duration of an utterance. This information is available in the J.P.L. system, but it is not included in the final parameterization of the word. This decision was made in the attempt to keep such voicing characteristics from affecting the recognition strategy. If this property were to be used, the rate at which a word was spoken, (i.e. the intraword pacing), would exert an influence upon the later comparison measures.

The VOice Feature EXtraction (VOFEX) hardware consists of four identical pairs of circuits, (details of these circuits are presented in appendix A). Each pair is connected to a different bandpass filter output, and is comprised of a zero-crossing circuit and an independent energy averaging circuit (figure 3.2.2). The zero-crossing values provide frequency distribution information in each range over the length of the word. The energy measures give an indication of energy concentration between the four bands during a given utterance segment. This results in a "two-dimensional" word description.

figure 3.2.2
Hardware (VOFEX) Supervised Feature Extraction



The four bandpass filters used each have an 18 db per octave maximum rolloff rate. In simpler terms, frequencies above and below the band settings are not attenuated completely, but are reduced in proportion to their distances from the filter settings. As a result of this filter property, in actual performance the bandpass filters could not provide a means for gathering data completely from within one formant range. The speech waves data collected was somewhat dependent upon the higher amplitudes of the lower frequencies. At later stages in the feature extraction implementation, the initial filter settings were therefore adjusted to provide for narrower bandpasses. This adjustment was intended to help in the achievement of better formant independent data collection. This partial solution to this problem also entailed the raising of the hysteresis of zero-crossing detection circuit in the VOFEX. A further solution would involve the purchasing or building of higher order bandpass filters. (The final filter settings are listed in appendix B).

The zero-crossing counters are individually read and reset by the LSI-11 microprocessor by means of a DEC DRV-11 parallel interface board. The average energies are applied as inputs to an ADAC Corporation analog-to-digital converter board [ADAC 00] under software control of the LSI-11 microprocessor. They are reset(cleared) by means of the

parallel interface. The A-to-D converter cannot become saturated by long windowing periods or large signal values due to input scaling through means of the compressor/mixer and protective circuits in the VOFEX. The sampling of the VOFEX outputs, and the triggering of the A-to-D converter are coordinated and controlled by an MDB KW11P programmable clock board in the LSI-11 microcomputer.

The hardware was designed to provide raw zero-crossing counts and non-normalized energy measures. In proceeding in this manner, the recognition system is not bound to their output representation. Different functions or measures can be developed to evaluate and represent zero-crossing information and can be implemented in software. This flexibilty is also present in the energy measure domain. This minimizing of the responsibility of the hardware helped keep the VOFEX construction costs low. The VOFEX hardware design, specifically its use of digital values for zero-crossing counts and windowing period controls, differs from the more constrained feature extraction methods previously used.

3.3 Data Compression and Normalization

The feature extraction process passes along to the remainder of the word recognition system eight words of information (the zero-crossing count and the average energy

for each of the four frequency bands) every ten milliseconds. Using a duration estimate of one second per utterance, 800 words of storage would be required to hold a description of each word in the vocabulary, if they were to be represented in the data form received from the VOFEX. A vocabulary of 100 words would take up approximately four times the storage available in the speech LSI-11 microcomputer. The form of the parameterization of a voiced utterance also has an effect upon the design and performance of the comparison/classification process. A decision as to the identity of a spoken word is made on the basis of how it best matches a word prototype in the vocabulary. The performance of such a decision mechanism is determined by the complexity of the comparison operation, and by the number of comparisons it is required to make. A comparison function which evaluates the similarity between two 800 word utterance parameterizations will be more complex and will require more execution time than one being used upon a more compact speech representation. The processes of reducing this volume of descriptive data and of representing word parameterizations to aid in the later decision operations, are called data compression and data normalization respectively.

In the development of real-time or near real-time word recognition systems, data compression techniques sacrifice the information content of the speech signal for processing speed and ease of representation. Dixon and Silverman [DIXO 75, SILV 74] follow a philosophy of "minimal loss of information" and do not make this compromise. For a microprocessor based system, data must be reduced and be compactly, conveniently represented.

In recognition systems that utilize linear predictive coding methods for data collection, data compression is attained at the same time as feature extraction. The output of such feature extractors are LPC coefficients and residual errors. In many such systems, this resultant information is used to segment the time series representation of speech into probable phoneme groups (e.g. [BEEK 00, WOLF 76]).

Most speech input systems that are used in the recognition of connected speech, must in some way differentiate periods of silence from periods of unvoiced speech. It is by this decision that such systems can then direct themselves at the recognition of individual words often by identifying boundaries between voiced and unvoiced segments. Atal and Rabiner [ATAL 76] calculate the means and standard deviations of selected speech properties to "tune" this decision section of their recognition system. Connected speech recognition systems require feature

extraction processes which will supply sufficient information to enable these voiced-unvoiced-silence decisions to be made.

In an isolated word recognition system, the critical silence- unvoiced decision does not exist. Periods of silence can be identified by means of the length of an "unvoiced" segment. Along with this design simplification accompanies the requirement that individual commands spoken to an isolated word recognizer be separated by a minimum period of silence. The resulting speech input will sound unnatural due to this pacing. This presents no problem in the use of this voice input system; the J.P.L. robotics control vocabulary is comprised of isolated command words. The command vocabulary can be extended to include short phrases as long as the interword silence periods are minimized during their voicings.

Gold [GOLD 66] points out that speech data does not fit into predetermined formats such as a Gaussian model of a proper dimension; Martin [MART 76] adds that no general mathematical theory exists which can preselect the information bearing portions of the speech data. The design of a recogniton system must incorporate heuristic and ad hoc strategies to enable its proper operation. It is in the data compression and normalization stages that many recognition systems whose feature extraction processes

appear similar diverge in order to achieve their respective final parameterizations.

Each word in the vocabulary has a fixed form(parameterization). The unknown input utterance will be compared with these prototypes; a classification is made based upon a best-match algorithm (presented in section 3.4). Before these comparisions can be made, the voice input must be represented in the same form as the known prototypes. A time dependent representation of the speech signal is used based upon the zero-crossing and energy information supplied by the VOFEX.

As noted earlier, eight words of information are arriving at the DEC LSI-11 microprocessor every ten milliseconds. The first method used to minimize buffer requirements and to keep speech processing to a minimum is to discard data samples representing a silence state. This decision is very similar to that of identifying the end of an utterance and enables the microprocessor to hold in storage VOFEX outputs describing only the voiced input. Rabiner and Sambur [RABI 75] present an algorithm for detecting the endpoints of isolated utterances of speech in a background of noise; it is based upon zero-crossing rate and energy. Their algorithm incorporates the calculation of statistical properties of the signal in the setting of its thresholds for the silence-nonsilence decision. These

computations require processor time in the attempt to achieve this speaker independent, self-adapting characteristic.

The properties made use of in the decision algorithm of the J.P.L. system are similar to those used by Rabiner and Sambur. It does not however use statistical measures in its operation. The beneficial self-adapting nature of their procedure is offset by the complexity of its implementation on the LSI-11 microprocessor and by the application of this system. Speaker dependent characteristics that will influence the silence decision can be stored with that speaker's vocabulary file, (see appendix D), in the form of threshold parameters for the recognition system. By proceeding in this manner, minimum values can be assigned(preset) for detecting the zero-crossing rates and energy levels for the four frequency bands which together represent speech. The only "learning" period required in the J.P.L. system for the identification of a silence state is ten milliseconds at the beginning of the recognizer operation to measure and record room noise levels.

In another recognition system, Paul [PAUL 70] uses only amplitude information in his endpoint decisions. In environments where a high signal-to-noise ratio exists, an effective algorithm can be developed based upon energy (level) values alone, specifically in the lower frequency

range. In less ideal environments, zero-crossing information can help in distinguishing weak fricative sounds from background noise. The threshold values used in the J.P.L. system were set experimentally after sampling the VOFEX outputs for "silence" segments. In evaluating the performance of the word detection routine, it was found that the zero-crossing information was not consistent enough in character to be used in the utterance start-stop algorithm. The bandpass filters were not providing sufficient attenuation of frequencies outside their ranges, and therefore, unwanted amplitudes were affecting the detection of zero-crossings at unvoiced-voiced boundaries. The J.P.L. endpoint algorithm was modified to utilize only the amplitude information provided by the four frequency bands in its decision making.

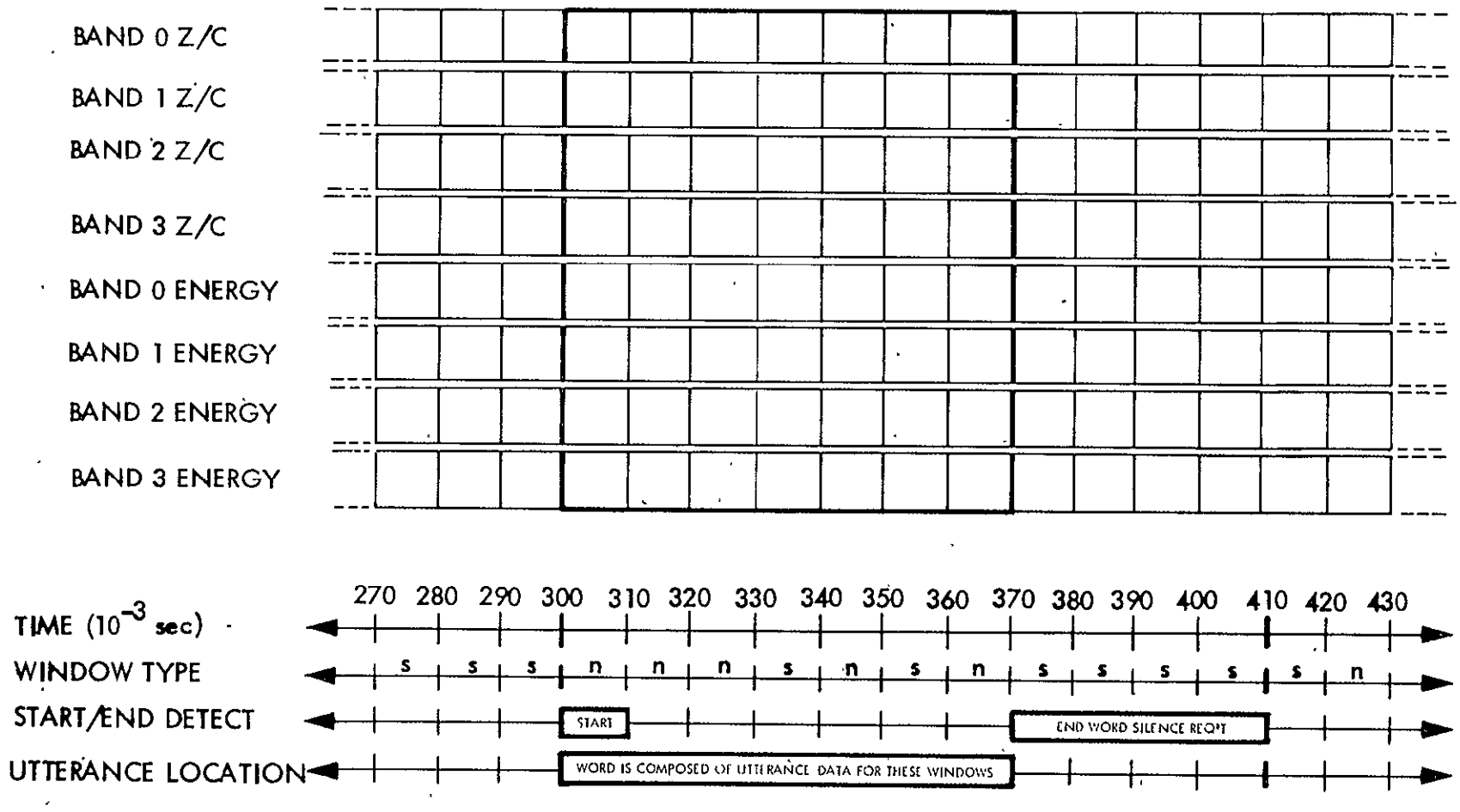
The start and end of an utterance is not abrupt; speech is continuous not discrete in nature. Due to the presence of noise and unvoiced segments, a recognition system cannot look at a single ten millisecond window and make a decision as to the location of an endpoint. A more reasonable and accurate algorithm would require a certain period of non-silence to indicate the start of a word, and thereafter, a definite period of silence would signify the end of the word. Initially these values were chosen to be four and five window lengths (40 and 50 milliseconds)

respectively. One needs such durations to insure that a burst of noise does not trigger the recognizer, and that an unvoiced segment within an utterance does not terminate prematurely the collection of data.

As the result of implementation considerations, the word detection algorithm used requires only one window of non-silence to indicate the start of a word. False starts are detected and discarded by imposing a minimum utterance length upon the word. This length was initially set at eight window lengths (80 milliseconds) and extended after process evaluation (see appendix B).

The utterance is represented by the data collected from the beginning of the non-silence detection period until the beginning of the silence detection period. This makes maximum use of early low-level word voicings while tending to ignore less important trailing sound. Figure 3.3.1 illustrates the initial representation of an utterance by the recognition routines.

A maximum utterance duration is enforced for implementation considerations (buffer size) and as a system precaution. The system is initially set to halt utterance data collection three seconds after the beginning of speech is detected. This can be easily changed without affecting the operation of the recognizer software and places little



WINDOW TYPE: s = SILENCE
n = NON-SILENCE

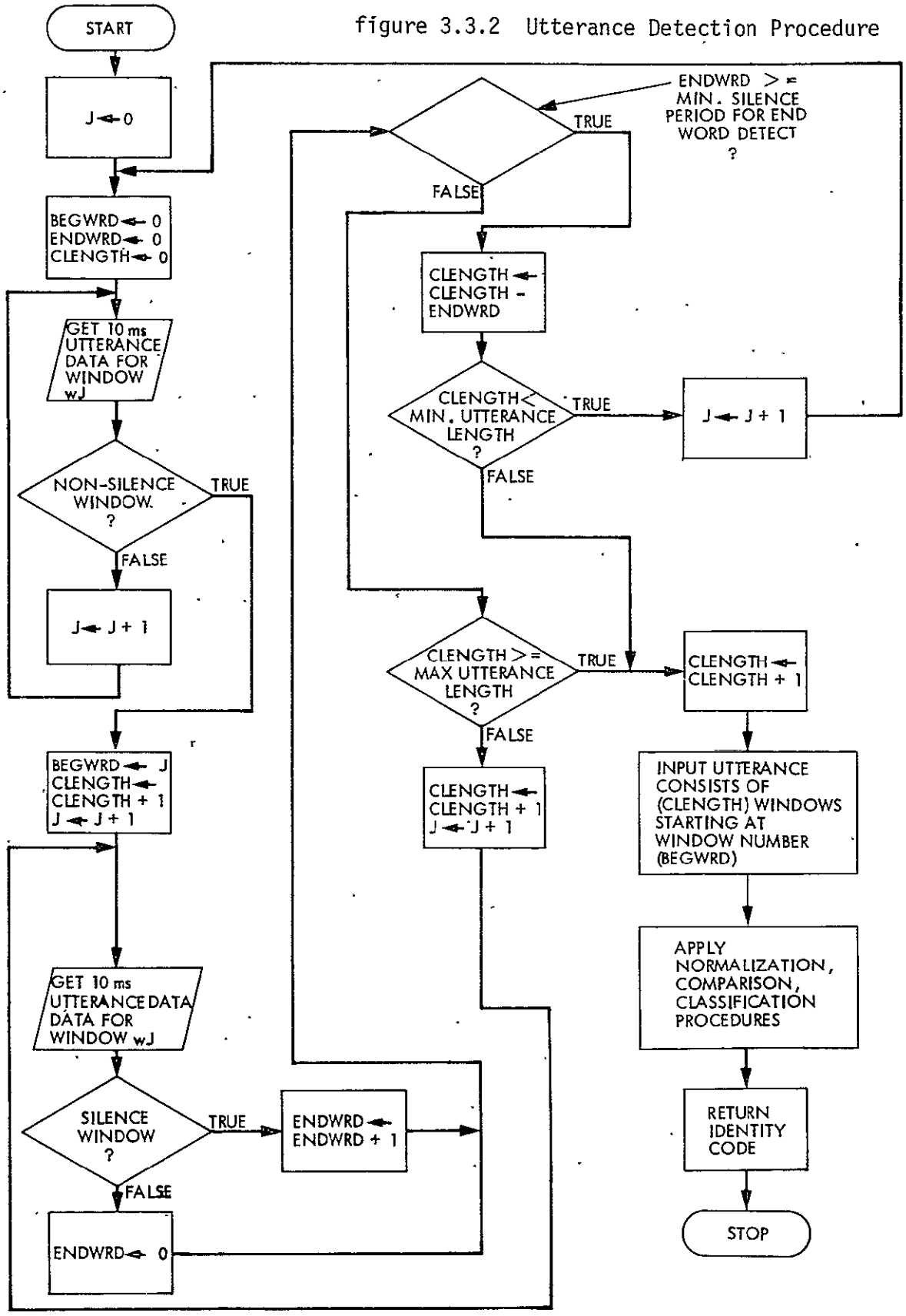
figure 3.3.1 Utterance Location Based Upon Non-silence Detection

constraint upon the composition or vocalization of the vocabulary. This value would need to be altered to permit the addition of longer input words to the vocabulary or the recognition of inputs from a very slow speaking operator. (Figure 3.3.2 represents the detailed utterance detection procedure used in the J.P.L. system).

Once the data for an utterance has been collected, data compression and normalization routines must be applied to achieve a representation in the same form as the vocabulary prototypes. The parameterized form chosen for the J.P.L. recognition system was influenced by earlier speech recognition approaches, but was principally developed as a result of the types of signal values being measured.

Bobrow and Klatt [BOBR 68] in their isolated word recognition work use "property extractors" to evaluate speech features and then apply functions based upon these extractors to reduce the range of their values. They chose to represent the speech input on the word level. Some systems which have been developed use a phoneme level representation of speech. This requires additional collection and processing of information to segment the word into the phoneme groups. Phoneme segmentation decisions are subject to considerable error; phoneme connecting rules are used in some systems to aid in error correction [ITAH 73]. Phoneme segmentation algorithms are characteristic of

figure 3.3.2 Utterance Detection Procedure



connected speech recognition and speech understanding systems. Such an approach is not needed in the J.P.L. system because of the isolated word nature of the input and the proportion of mono-syllabic utterances in the vocabulary.

Nearly linear scalings of data are used by Paul [PAUL 70] and by Neroth [NERO 72] in their systems. Paul achieves a standard length data representation of an input by discarding data from within the utterance ("shrinking" the vector) or by introducing redundant values ("stretching" the vector.) Neroth represents his utterance by segmenting his list of feature values of zero-crossings and amplitudes into seven near equal in duration measurement groups. By similarly dividing utterances into a number of feature periods, and by computing representative zero-crossing rates and average energies for each of the four bands for each segment duration, a reasonable compromise between accuracy (correct word identification), and storage and processing speed (near real-time) can be realized.

To segment the utterance data, a linear time-normalizing procedure was chosen. An averaging technique is then applied to the individual component "windows" to arrive at representative values for zero-crossings and energies for each speech interval. A strategy of segmentation which results in sixteen utterance

divisions is used; this representation requires 128 values in the encoding of each word in the vocabulary. In the initial system, a segmentation algorithm was used which resulted in eight utterance divisions. This compression value produced utterance parameterizations in which much valuable data was reduced to the point of being uninformative. This situation necessitated the choice of a larger segmentation constant. The segment value sixteen enables short computations based upon representative utterance encodings to be programmed and executed.

Data "stretching" is required when an utterance is detected which is less than sixteen window segments (160 milliseconds) in duration. This operation would be used upon features passed to the software recognizer which have been extracted from sounds too short to result from an input word. For this reason, the J.P.L. recognition system considers such data as resulting from noise and discards it.

Utterance "shrinking" is required when the detected utterance is longer than sixteen window segments in duration. Linear(equal) "shrinking" will uniformly compress the speech data. This is desired if one does not want signal information collected during specific events (e.g. start of utterance, end of utterance, phoneme transition) to be overly represented in the parameterized word sample. In the design of the J.P.L. system, the responsibilty for

stressing such features lies in the comparison/classification routines. The output of the data compression/normalization section is a uniform speech sample which provides the opportunity to later locally test and implement a variety of decision methods.

The following segmentation algorithm is used by Neroth to calculate the number of window samples to incorporate together for each of his normalized utterance sections. L is the length of the utterance in units of "windows." N is the number of sections that the utterance is to be segmented into, which will be refered to as the "segmentation number." Neroth uses the value seven for N. His ith section is composed by averaging the values of the K(i-1)+1, K(i-1)+2, K(i-1)+3, ..., K(i) data windows for i=1, 2, 3, ..., N.

$$K(i) = K(i-1) + s + r \quad \text{for } i=1, 2, 3, \ldots, N \text{ where}$$

$$K(0) = 0 \text{ by definition,}$$

$$s = \lfloor L/N \rfloor,$$

$$r = 1 \quad \text{if } L-(s*N) >= i$$
$$r = 0 \quad \text{otherwise}$$

close inspection of this segmentation algorithm will show that its non-linearity causes the parametric representation of the utterance to stress properties detected near its end. For example if L=18 the following divisions are computed (windows are labeled sequentially by their number; vertical

bars illustrate segmentation points):

| 1 2 3 | 4 5 6 | 7 8 9 | 10 11 12 | 13 14 | 15 16 | 17 18 |

A more uniform segmentation would appear as:

| 1 2 3 | 4 5 | 6 7 8 | 9 10 | 11 12 13 | 14 15 | 16 17 18 |

The following algorithm is proposed to accomplish it:

$$Z(0) = 0$$

$$Z(i) = \lfloor ((i*L)/N) + 0.5 \rfloor \quad \text{for } i=1, 2, 3, \ldots, N$$

The function name "Z" is used instead of "K" to differentiate this new algorithm from that used by Neroth. It is computationally simple and easily implemented through assembly language instructions on the DEC LSI-11 microprocessor. Both the "K" and "Z" segmentation methods are approximations to the ideal routine which would use equal utterance intervals of (L/N) "windows." (These approximations are valid for utterances of a reasonable duration.) The ideal method would necessitate the use of non-integer length window sections and would require greater processing complexity than either the "K" or "Z" method. The "Z" segmentation method is used in the compression/normalization procedure for the J.P.L. speech recognizer process.

Using the "Z" method for calculating the sixteen time sections of the utterance, an averaging scheme can now be applied to the zero-crossing and energy data that describes the set of "windows" comprising each segment. Using a speech utterance sample with a length (L) of 36 and a segmentation number (N) equal to sixteen, the following segmentation is computed:

```
| 01 02 | 03 04 05 | 06 07 | 08 09 | 10 11 |
        | 12 13 14 | 15 16 | 17 18 |
| 19 20 | 21 22 23 | 24 25 | 26 27 | 28 29 |
        | 30 31 32 | 33 34 | 35 36 |
```

The actual number of windows that comprise each segment for the above example(L=36, N=16) is:

```
2  for segments  1,3,4,5,7,8,9,11,12,13,15,16  and
3  for segments  2,6,10,14
```

The following standardized vector "V" is achieved by reducing the data by means of averaging the information contained in each segment:

```
| V(1) | V(2) | V(3) | ... | V(16) |
```

Formally, the "V" vector is computed for a single set of data at a time, (e.g. the zero-crossings for band 0, the energies for band 3, etc.); there are eight sets of "V"

vectors, (zero-crossings and energies for each of the four bands). If we use D(j) to represent the data value of the ith window for a specfic class of information, i=1, 2, 3, ..., L, (utterance of length L), then "V" is calculated by the following procedure:

$$V(i) = \left\lfloor (1/(Z(i)-Z(i-1))) \sum_{k=Z(i-1)+1}^{Z(i)} D(k) \right\rfloor$$

for i=1, 2, 3, ..., N

and by definition Z(0)=0

By using this method for each class of utterance information, the word vector form illustrated in figure 3.3.3 is achieved. Notice that at this point in compression/normalization, the utterance continues to be represented by raw data. If the classification process used a decision system based solely upon the similarities of such signal measures, this form could be used to store the vocabulary prototypes and to represent the unknown word.

The properties of zero-crossing rate and average energy were chosen to represent features descriptive of human speech. It is the relative zero-crossing values within a given band that is representative of the evolution of the principle frequency components. Raw zero-crossing values

SEGMENTATION OF UTTERANCE WINDOWS (L = 18 and N = 7)

BAND 0 Z/C	2	3	3	5	4	5	5	6	5	4	2	6	2	2	0	0	1	2
BAND 1 Z/C	10	11	10	8	7	8	7	9	12	11	15	14	13	14	14	12	8	8
BAND 2 Z/C	18	19	18	19	21	24	28	27	23	20	18	19	20	26	26	24	22	21
BAND 3 Z/C	28	31	32	38	40	40	36	35	40	40	38	37	35	35	35	32	31	34
BAND 0 ENERGY	210	250	261	288	401	381	411	401	382	370	298	312	313	350	371	360	214	203
BAND 1 ENERGY	401	441	390	370	301	271	282	308	425	470	507	515	513	488	512	550	552	540
BAND 2 ENERGY	1000	1012	1400	1311	1398	1251	1259	1201	1311	1517	1701	1880	1750	1670	1590	1409	1371	1308
BAND 3 ENERGY	201	247	244	301	399	390	381	383	390	371	347	344	351	330	290	271	277	241

BAND 0 Z/C	2	4	5	4	3	1	1
BAND 1 Z/C	10	7	8	11	14	14	9
BAND 2 Z/C	18	20	26	21	19	26	22
BAND 3 Z/C	30	39	37	40	36	35	32
BAND 0 ENERGY	240	344	397	376	308	360	259
BAND 1 ENERGY	420	335	287	447	511	500	547
BAND 2 ENERGY	1070	1354	1237	1414	1777	1630	1362
BAND 3 ENERGY	230	350	384	380	347	310	263

COMPRESSED WORD VECTORS OF RAW DATA

figure 3.3.3
Compressed Word Vector Consisting of Raw Data

are not as compact or informative as are functions of these rates that have been developed. Niederjohn [NIED 75], presents a variety of different techniques that have been useful in extracting significant features from zero-crossing data. One such processing of zero-crossing information is the calculation of the number of time intervals for which zero-crossing rates are between two values. In the J.P.L. system, the representation of such time dependent characteristics is accomplished by ranking the utterance intervals based upon their zero-crossing counts. Four separate rankings of zero-crossing values for the length of the standardized utterance is used, one ranking for each band. This is easily calculated by means of a sorting procedure applied to the vector elements; this method requires less computation and software than many other zero-crossing techniques. In using this normalization method upon the zero-crossing values averaged for a single band (values are represented in the previously defined vector "V"), the ranked zero-crossing measures are represented in vector "RZ." "RZ" exhibits the following element relationship:

$$\forall_{i,j} \; i,j=1, 2, 3, \ldots, N$$
$$RZ(i) > RZ(j) \Rightarrow V(i) > V(j)$$

and RZ is obtained by reordering and averaging the values:

| 00 | 02 | 04 | 06 | 08 | 10 | ... | 2*(N-1) |

For example, the "V" vector (with N=8):

| 12 | 15 | 11 | 08 | 09 | 14 | 16 | 19 |

would be represented as the "RZ" vector:

| 06 | 10 | 04 | 00 | 02 | 08 | 12 | 14 |

and, the "V" vector (with N=8):

| 12 | 15 | 15 | 08 | 09 | 09 | 16 | 15 |

would be represented by the "RZ" vector:

| 06 | 10 | 10 | 00 | 03 | 03 | 14 | 10 |

If raw energy values are used in the final normalized speech representation, the recognition system will be highly sensitive to voicing characteristics whose use will provide misleading information to the identification process, (e.g. speaking level, proximity to microphone). What is needed by the classifier is some measure of the relative differences between the energy distributions found within utterances. Before different utterance parameterizations can be compared, the energy data in the two words must be represented in identical ranges. Neroth [NERO 72] and Reddy [REDD 67] both normalize their energy measures by dividing all amplitude data in a given sample by its maximum amplitude. Paul [PAUL 70] uses a procedure which divides

his sampled amplitudes by their average amplitude value. All of these methods generate fractional results which must be stored and carried along through the remainder of the recognizer system. Fractional arithmetic requires more processing time than does integer. The computational capabilities available in the LSI-11 microcomputer are very limited; it is for this reason that the algorithms used in the J.P.L. recognition system are designed for and implemented with integer values.

To normalize the 64 standardized energy measures for an entire input utterance (sixteen measures for each of the four bands), the data in each band is represented as an offset from the minimum value, scaled and then divided by the range of energy values for the given band. This method yields an integer result of maximum significance for the LSI-11 microprocessor word size (16 bits including sign). The values calculated by this procedure should better reflect changes in the amplitude measures than the algorithm used by Reddy. A disadvantage of this method is that it will produce unreliable and misleading results for values over a small range. To guard against this occurrence, the amplification circuits in the VOFEX have been tuned to the characteristic amplitudes of speech passed through each bandpass filter to provide proper amplitude ranges. The procedure for generating the normalized energy vector "EV"

for a given band of the utterance from the "V" standardized raw energy vector is:

MAXEN = maximum energy value of the N energy samples in the utterance band

MINEN = minimum energy value of the N energy samples in the utterance band

EV(i) = ((V(i) - MINEN) * 32,768) / (MAXEN - MINEN + 1) for i=1, 2, 3, ..., N

The final normalized form for a given utterance is illustrated in figure 3.3.4. Data has been reduced from that shown by figures 3.3.1 and 3.3.3. The feature extraction and data compression/normalization processes have been designed to supply a concise, robust utterance representation to the decision process. This enables the comparison/classification routines to evaluate the identity of a speech input rapidly and accurately. Plots of the input utterances "ROOT" and "TERRAIN" are displayed in figures 3.3.5 and 3.3.6 respectively; the plots were made using utterance data in the final normalized form.

3.4 Utterance Comparison and Classification

The feature extraction and the data compression/normalization routines pass along to this final recognition system process a compact description of the input utterance in the form of one "RZ" vector and one "EV"

figure 3.3.4
Compressed Word Vector Consisting of Normalized Data

COMPRESSED WORD VECTORS (N = 7)

BAND 0 Z/C	2	4	5	4	3	1	1
BAND 1 Z/C	10	7	8	11	14	14	9
BAND 2 Z/C	18	20	26	21	19	26	22
BAND 3 Z/C	30	39	37	40	36	35	32
BAND 0 ENERGY	240	344	397	376	308	360	259
BAND 1 ENERGY	420	335	287	447	511	500	547
BAND 2 ENERGY	1070	1354	1237	1414	1777	1630	1362
BAND 3 ENERGY	230	350	384	380	347	310	263

NORMALIZATION

BAND 0 Z/C	4	9	12	9	6	1	1
BAND 1 Z/C	6	0	2	8	11	11	4
BAND 2 Z/C	0	4	11	8	2	11	6
BAND 3 Z/C	0	10	8	12	6	4	2
BAND 0 ENERGY	0	2156	32560	28205	14102	24886	394
BAND 1 ENERGY	16698	6026	0	20088	28123	26742	32643
BAND 2 ENERGY	0	13144	7728	15920	32719	25916	13513
BAND 3 ENERGY	0	25369	32557	31711	24734	16912	6976

COMPRESSED WORD VECTORS OF NORMALIZED DATA

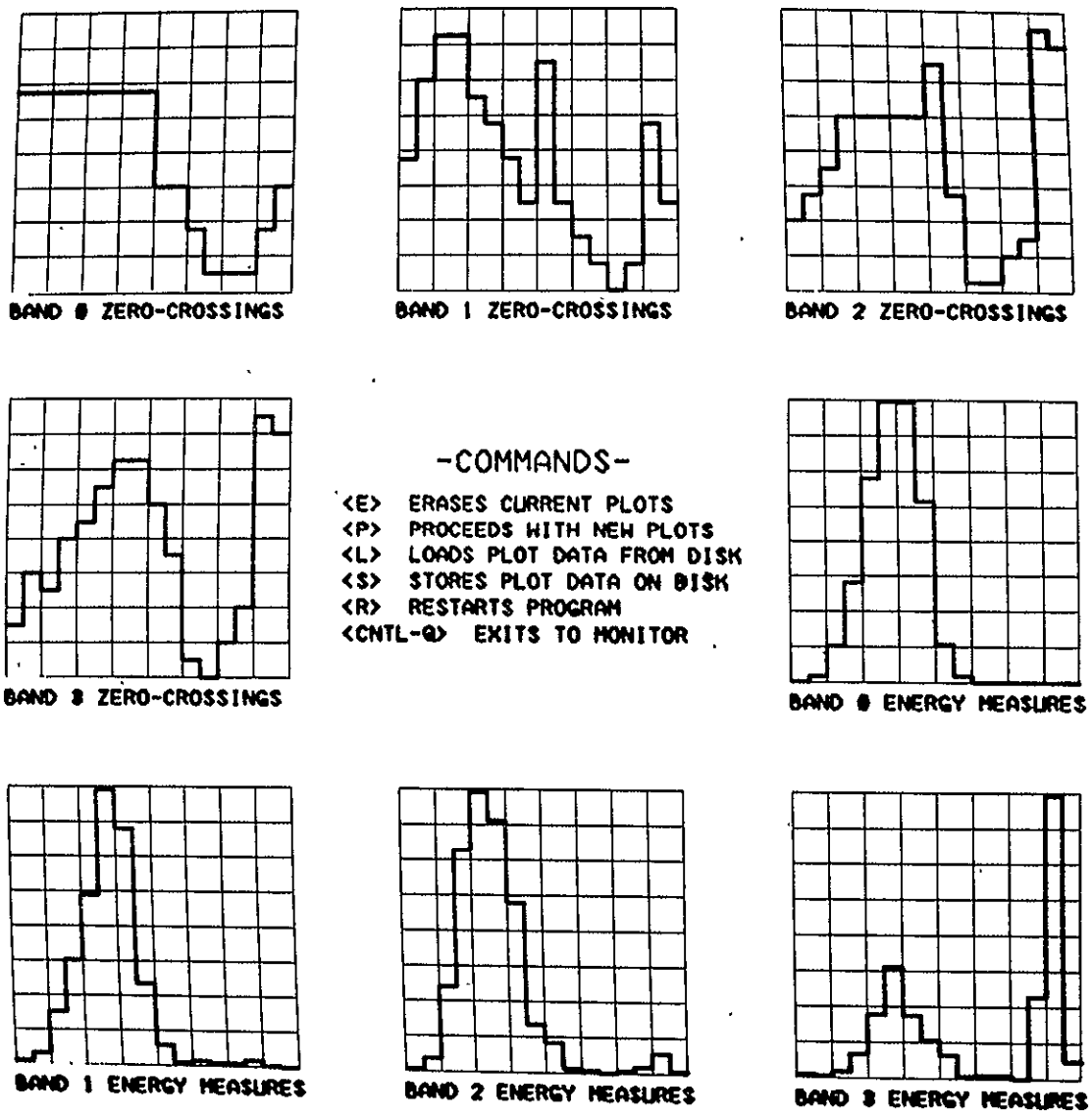


figure 3.3.5 Plot of Normalized Data for Command "ROOT"

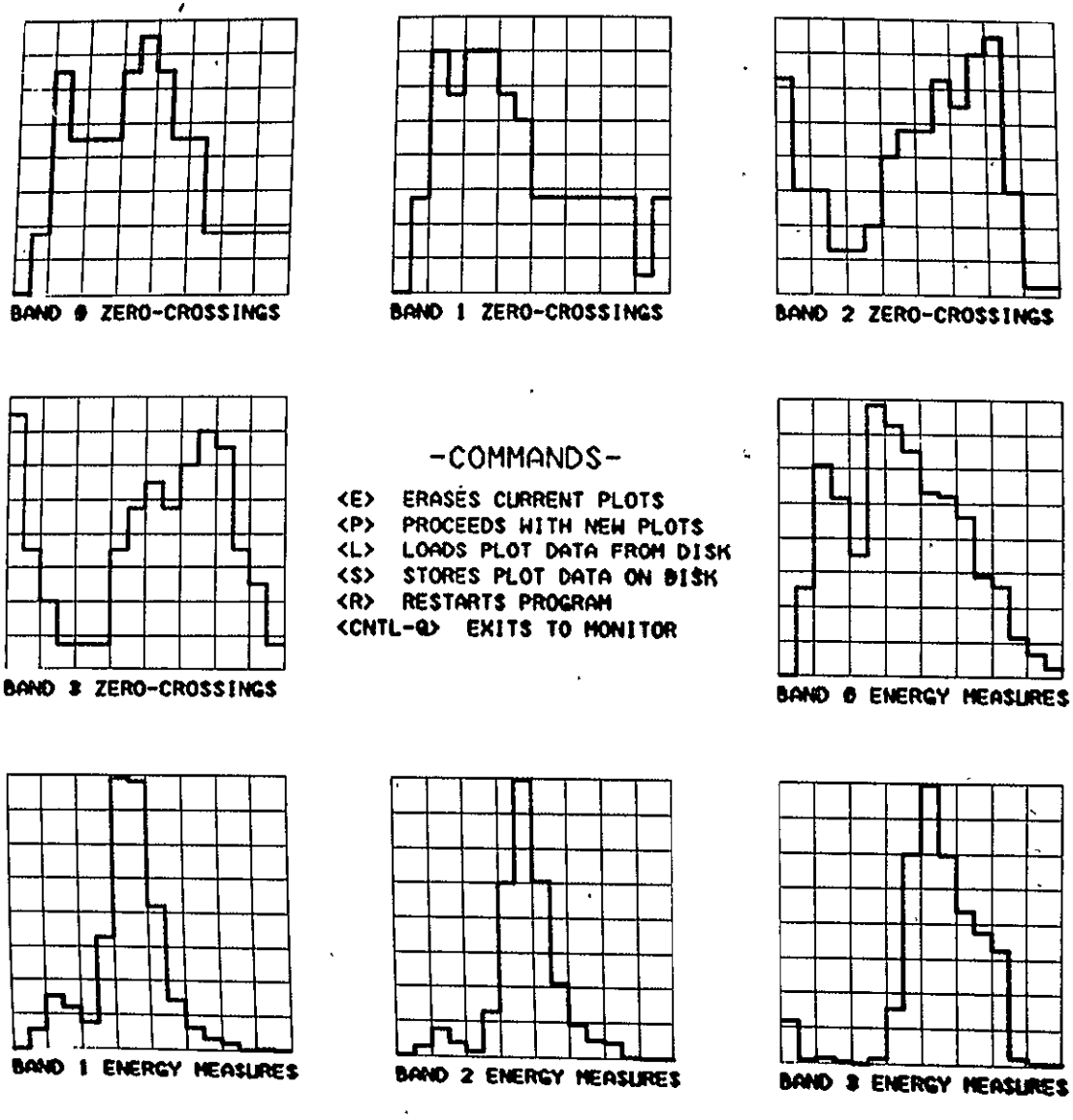


figure 3.3.6 Plot of Normalized Data for Command "TERRAIN"

vector for each of the four frequency bands. The input utterance is represented by a parameterization requiring 128 words of storage. (For each word prototype in the vocabulary file, only 64 words of storage are used as the result of a further reduction step). On the basis of the similarities of the unknown input to the known vocabulary, the comparison/classification process selects the most likely word identity. Different similarity measures and classification strategies provide different tradeoffs between accuracy and speed. Heuristics are often included in systems to aid in their recognition performance.

The unknown input word must in some way be compared with each reference pattern to determine to which of the reference patterns it is most similar. In other recognition systems, this similarity has been based on the minimum distance or the maximum correlation between a reference pattern and the unknown utterance, where a pattern sample is treated as an N-element vector. Two commonly used distance measures are the Euclidean [ATAL 72, PAUL 70] and the Chebyshev [BOBR 68, MCDO 00, NERO 72] norms. To illustrate the difference between these measures, two N-element vectors A and B are used.

Euclidean distance:

$$ED(A,B) = \sqrt{\sum_{j=1}^{N} (A(j)-B(j))^2}$$

Chebyshev distance:

$$CD(A,B) = \sum_{j=1}^{N} | A(j)-B(j) |$$

The Euclidean measure is computationally more complex than the Chebyshev as squaring operations are required, (the square root is not necessary as in a minimum distance classification, the performance of a Euclidean squared measure is identical to that of a Euclidean measure).

Often poor recognition performance results from improper detection of the beginning or end of an utterance [REDD 67]. This problem has been treated at the comparison/classification stage by two methods: dynamic programming [HATO 74, ITAK 75, LOWE 76, NIPP 76, WOLF 76] and vector element shifting. Dynamic programming is a non-linear time normalization technique. It is often used in recognition systems which utilize linear predictive coding feature extraction. Its usefulness lies in its ability to align critical points (e.g. peaks, inter-syllable minimums) when comparing two parameterizations. This pattern sample warping achieves

better interior matching (especially of multi-syllabic words) than a linear time-normalization procedure. Dynamic programming can be used in conjunction with both the Euclidean and Chebyshev distance measures.

In section 3.3, reasons were presented for the choice of a linear time normalization method for the J.P.L. recognizer. Linear time scaling shrinks utterances to the standard sixteen segment length. This technique will cause the utterance representation to be sensitive to the speaker's intraword pacing characteristics. Interior mismatch between an unknown utterance and a pattern sample will affect the accuracy of the comparison operation. This performance degradation will be least for mono-syllabic inputs as there exist fewer points at which their voicing rates can change. White [WHIT 76a] has found that linear time normalization with left and right shifting is "as good as" dynamic programming in the recognition of mono-syllabic utterances.

This shifting method of comparison is used in the classification process. The distance between two utterances A and B, using a Chebyshev norm is represented by the value SCD:

$$SCD(A,B) = \min(CDL(A,B), ((N-1)/N)*CD(A,B), CDR(A,B))$$

where $$CDL(A,B) = \sum_{j=1}^{N-1} D(j+1,j)$$

$$CDR(A,B) = \sum_{j=1}^{N-1} D(j,j+1)$$

$$D(i,j) = | A(i)-B(j) |$$

CDL(A,B) and CDR(A,B) are the Chebyshev distances between vectors A and B with vector A shifted one element to the left and to the right respectively. The value ((N-1)/N) is used to adjust for the summation of N-1 terms in the shifted comparison measures and N terms in the non-shifted CD(A,B) calculation.

In computing the total distance between two word pattern samples in the J.P.L. system, eight SCD computations are performed and accumulated, (distances for the zero-crossings in band 0, for the energies in band 3, etc.). The total shifted Chebyshev distance between pattern sample PS1 and pattern sample PS2 is called TSCD and is defined as:

```
TSCD(PS1,PS2) = SCD(PS1 RZ band 0,PS2 RZ band 0)
              + SCD(PS1 RZ band 1,PS2 RZ band 1)
              + SCD(PS1 RZ band 2,PS2 RZ band 2)
              + SCD(PS1 RZ band 3,PS2 RZ band 3)
              + SCD(PS1 EV band 0,PS2 EV band 0)
              + SCD(PS1 EV band 1,PS2 EV band 1)
              + SCD(PS1 EV band 2,PS2 EV band 2)
              + SCD(PS1 EV band 3,PS2 EV band 3)
```

In word parameterizations, the value range and information content of all elements are usually not equivalent. For example, the zero-crossing ranks are values from 0 to 2*(N-1), but the energy values are represented by 15-bit numbers. Information supplied by the zero-crossing rank for band 0 might not prove as helpful in making a recognition decision as the energy value of band 0 or band 3. For these reasons, a weighted distance measure is utilized in the comparison/classification process of the J.P.L. system. The total weighted shifted Chebyshev distance between pattern sample PS1 and pattern sample PS2 is called TWSCD and is calculated as:

```
TWSCD(PS1,PS2) = wz(0)*SCD(PS1 RZ band 0,PS2 RZ band 0)
               + wz(1)*SCD(PS1 RZ band 1,PS2 RZ band 1)
               + wz(2)*SCD(PS1 RZ band 2,PS2 RZ band 2)
               + wz(3)*SCD(PS1 RZ band 3,PS2 RZ band 3)
               + we(0)*SCD(PS1 EV band 0,PS2 EV band 0)
               + we(1)*SCD(PS1 EV band 1,PS2 EV band 1)
               + we(2)*SCD(PS1 EV band 2,PS2 EV band 2)
               + we(3)*SCD(PS1 EV band 3,PS2 EV band 3)

  where  wz(i) = the ith zero-crossing band weighting and
         we(i) = the ith energy band weighting
                 for i=0, 1, 2, 3
```

This comparison function is implemented in the PDP-11 assembly language and allows the development and evaluation of different decision criteria. Initially, the same weighting vectors are used for each speaker. However, different vectors can be utilized for different users as the weights are stored along with the speaker's voice characteristic variables and vocabulary. (See appendix D for sample weights).

Using the TWSCD formula, similarity measures of the unknown utterance to each of the stored vocabulary prototypes is computed. The values returned by this procedure represent distances between points in a vector space of 8*N, where N is the number of elements in each of the four RZ zero-crossing and four EV energy vectors. A perfect match of the unknown to one of the vocabulary words will yield a TWSCD value of zero. Progressively larger values indicate less similar parameterizations.

A common classification technique is to compare the input utterance to each stored prototype and select as the identity the one with the lowest distance(difference) measure. This exhaustive comparison is acceptable in recognizers having small vocabularies, but time-consuming in larger systems. As the vocabulary grows, the potential confusability between words increases (i.e. the vector space has finite domain and each word is represented by a

point in the space). Some procedure is required to achieve high recognition accuracy and speed in systems employing medium or large size vocabularies (greater than 60 words).

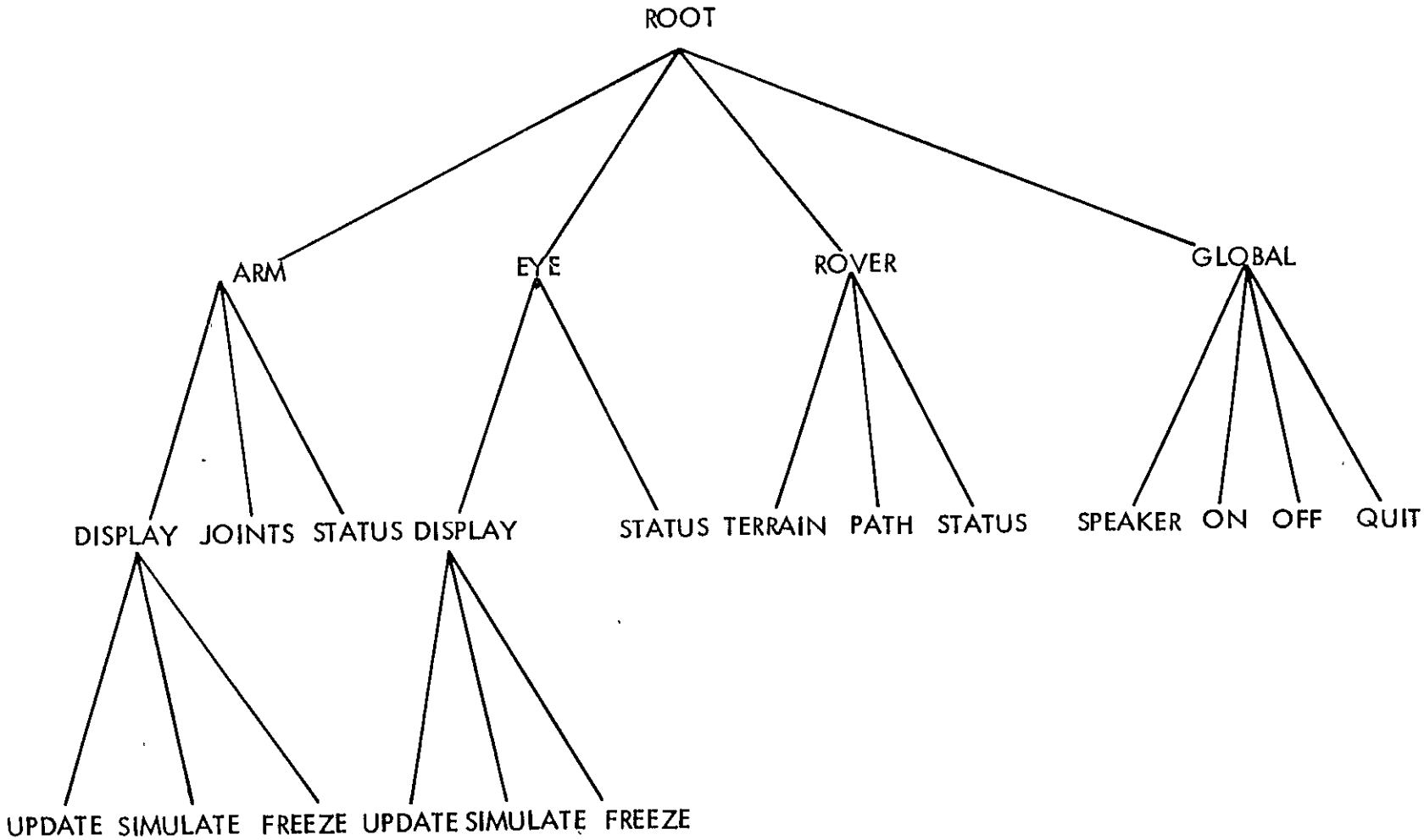
Neely and White [NEEL 74] suggest using the ratio of the second lowest score to the lowest as a measure of the confidence of the nearest-neighbor decision. Itakura [ITAK 75] rejects a reference pattern during matching if its distance from the input pattern sample is ever over a certain threshold. Warren [WARR 71] dynamically removes candidates from consideration as his system learns more about the input. Grammars have been utilized by Haton [HATO 74], and Neely and White [NEEL 74] in using syntactic analysis to help in the performance of their systems.

The J.P.L. system uses in its classification process a threshold upon the minimum distance found, a threshold upon a confidence measure similar to that by Neely and White, and a structured vocabulary to achieve its desired performance. The vocabulary of the current speaker consists of global and local commands (figure 3.4.1 for partial vocabulary, appendix C for complete vocabulary). The global commands are system commands affecting the domain and configuration of the recognizer, while the local commands are the actual instructions being dictated to the robotic systems.

LOCAL COMMANDS	GLOBAL COMMANDS
ROOT	GLOBAL
ARM	SPEAKER
DISPLAY	ON
UPDATE	OFF
SIMULATE	QUIT
FREEZE	
JOINTS	
STATUS	
EYE	
ROVER	
TERRAIN	
PATH	

figure 3.4.1
Partial Local and Global Command Vocabulary

The local and global commands are tree-structured (figure 3.4.2 for structure of partial vocabulary, appendix C for structure of complete vocabulary). This imposes syntactic constraints upon the input utterances. The user begins the session at the root of the tree from which only a subset of the vocabulary is available. The state of the recognizer is represented by the node at which the user currently resides. From a given local node, the available commands consist of: the root node, the current local node itself, an immediate descendent node, a brother(sister) node, an immediate ancestor node or the global subtree root node. From the global subtree root node, the available commands consist of the descendent global nodes. The parameterization of the input utterance is only compared with the prototypes of the available commands for the given curent state. This limited domain technique results in the



23 NODES
17 COMMANDS

figure 3.4.2 Partial Tree-structured Command Vocabulary

exclusion of comparison operations involving words in the vocabulary which are not within the current context of the system. This speeds up the recognizer comparison/classification process and improves system accuracy.

To insure that an undefined or "inaccessible" utterance was not input, two thresholding techniques are applied after the two "nearest" prototypes of the current vocabulary subset to the unknown word have been determined. The confidence of the best match is represented by the quotient which results from dividing the second smallest prototype distance by the smallest prototype distance. This value must exceed a given threshold to help insure that the pattern sample selected is a good choice relative to the other possibilities. The raw distance value of the input utterance to the best legal match must be less than another threshold value. This test keeps the system from selecting a legal prototype which is most similar relative to the other legal choices, yet poor in terms of absolutely matching the input. If no reference pattern meets both these criteria, the system returns a "no-match" response. "No-match" decisions do not effect the recognizer state. (Sample thresholds are provided in appendix D).

When a valid(accessible) node meets the two previous threshold tests, the recognizer returns the digital code representing the command identity of the node and updates its state if necessary. (In reality, the recognizer state is updated to the new node position only in cases where the new node has descendents; this reduces the number of commands needed to traverse subtrees and makes the input facility more convenient to use. Note that it is the digital code representing the command word which is returned, not a code describing the new node, as the same word can be represented by multiple nodes in different subtrees).

When the new node is the global subtree root node, the previous state of the recognizer is saved before being updated and additional constraints are imposed by the system. A command following the global subtree command must be a global operation request represented by a descendent node of the global subtree root. After its voicing, the corresponding digital code is returned, and the recognizer state is restored to the state that was occupied before the global subtree request was made. Since global commands can change the mode of the recognizer (e.g. select new speaker, turn audio input off), the recognizer program must have knowledge of the identities of these commands; the digital codes for global commands are provided in the speaker's

vocabulary file (appendix D).

The following sample session demonstrates the command choice constraints imposed upon the user by the syntax rules of the vocabulary illustrated in figure 3.4.2. The user begins the session in the ROOT state. The following commands are legal: ROOT, ARM, EYE, ROVER and GLOBAL. The command is ARM; the new state is ARM. The available commands are: ROOT, ARM, DISPLAY, JOINTS, STATUS, EYE, ROVER and GLOBAL. The command is STATUS; the state remains ARM; the available commands are unchanged. The next command given is PATH. PATH is an illegal command from this point; a "no-match" code is returned; the state remains ARM. The next command is ROVER; the new state is ROVER. The available commands are: ROOT, ARM, EYE, ROVER, TERRAIN, PATH, STATUS and GLOBAL. The command is GLOBAL; the old state (ROVER) is saved; the new state is GLOBAL. The valid commands are: GLOBAL, SPEAKER, ON, OFF and QUIT. The command is SPEAKER; a new user vocabulary file is loaded; the new state is ROVER (the restored local state). The available commands are again, ROOT, ARM, EYE, ROVER, TERRAIN, PATH, STATUS and GLOBAL. This continues until the global command, QUIT is given.

Figure 3.4.3 represents in flowchart form the comparison/classification operations in the J.P.L. speech recognizer. The near real-time recognizition was attained

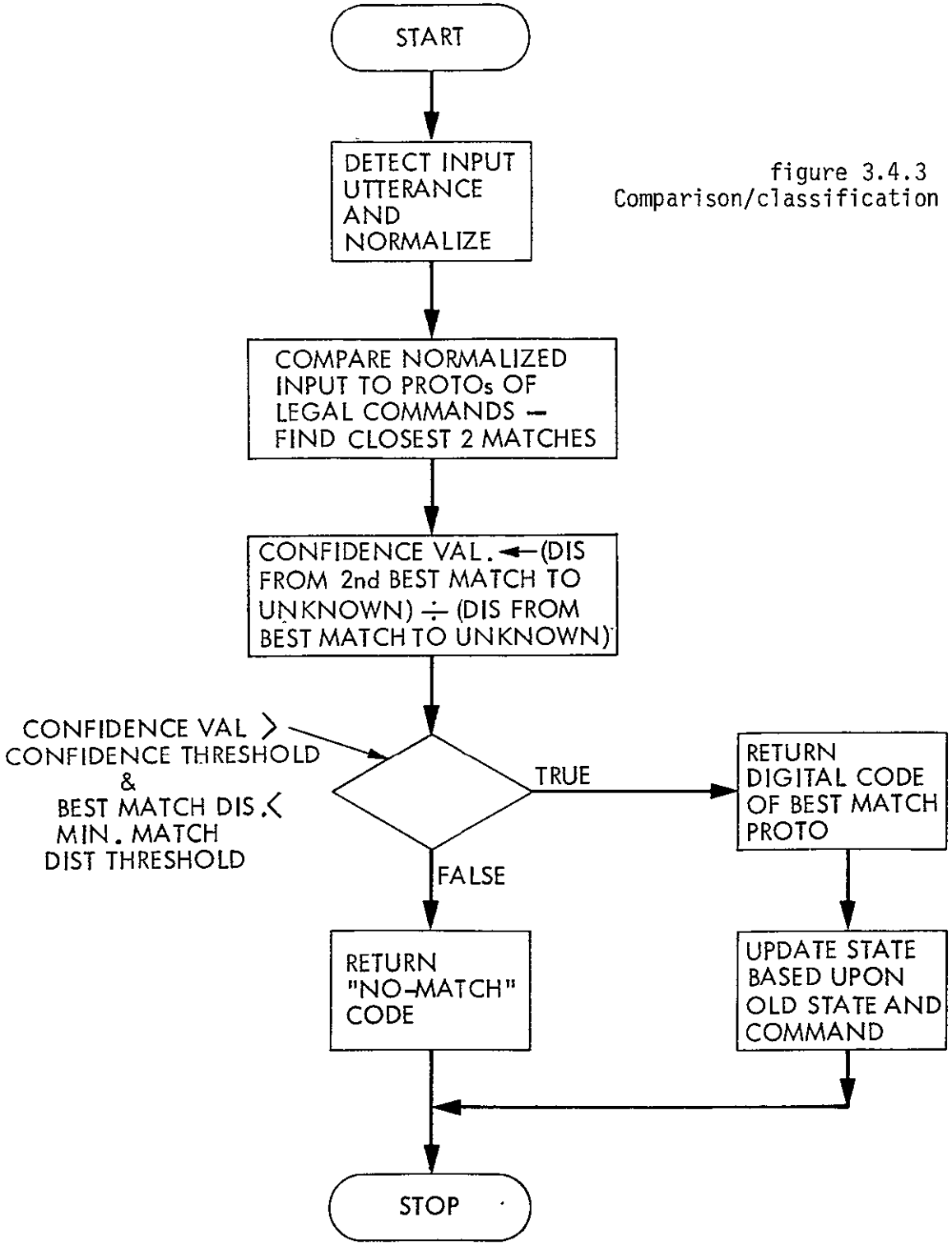


figure 3.4.3
Comparison/classification Procedure

by selecting and designing compression and matching algorithms which were compatible and microprocessor implementable. These procedures included linear time normalizing, Chebyshev norm distancing, utterance shifting and distance measure weighting which operated upon reference pattern samples from a vocabulary syntactically constrained.

3.5 Organization and Operation

Three software packages were developed to generate and supervise the speech input facility; these packages are VOCGEN, LEARN and RECOGNIZE. VOCGEN is comprised of the software routines which are responsible for transforming the user's vocabulary description, syntactic constraint rules and speaking parameters into the data forms required by the vocabulary training and word recognition systems. The user specifies the vocabulary in a hierarchical manner by means of listing node level values along with each command word identity and digital code. (In appendix C, a sample robotics application vocabulary specification is listed along with its corresponding VOCGEN execution summary). The digital code that is listed for each command word represents the identity of the command throughout the robot system. The syntactic constraint rules (presented in section 3.4) were developed for the J.P.L. robotics command application; however, different syntax rules could be used without

requiring any change in the underlying data structures of the speech input facility. VOCGEN produces as its output, a vocabulary description module which is stored on floppy disk.

The LEARNing program is used to generate the prototype set of a given user based upon the description module produced by VOCGEN. The user interactively voices examples of each word in the vocabulary. Prototype speech features are measured and recorded. Upon vocabulary learning completion, the user's trained vocabulary file is stored on floppy disk. This vocabulary file can be later recalled by means of the LEARN program to alter the stored word prototypes. It is this user trained vocabulary file which is used by the RECOGNIZE program.

The VOFEX hardware and RECOGNIZE software comprise the speech input facility and are responsible for the recognition of robot system commands. Following detection and recognition of an input utterance, RECOGNIZE sends the digital command code of the input to the communications subsystem, which then forwards it to the appropriate robot subsystem for execution. Figure 3.5.1 illustrates this interface of the speech input process to the robot system.

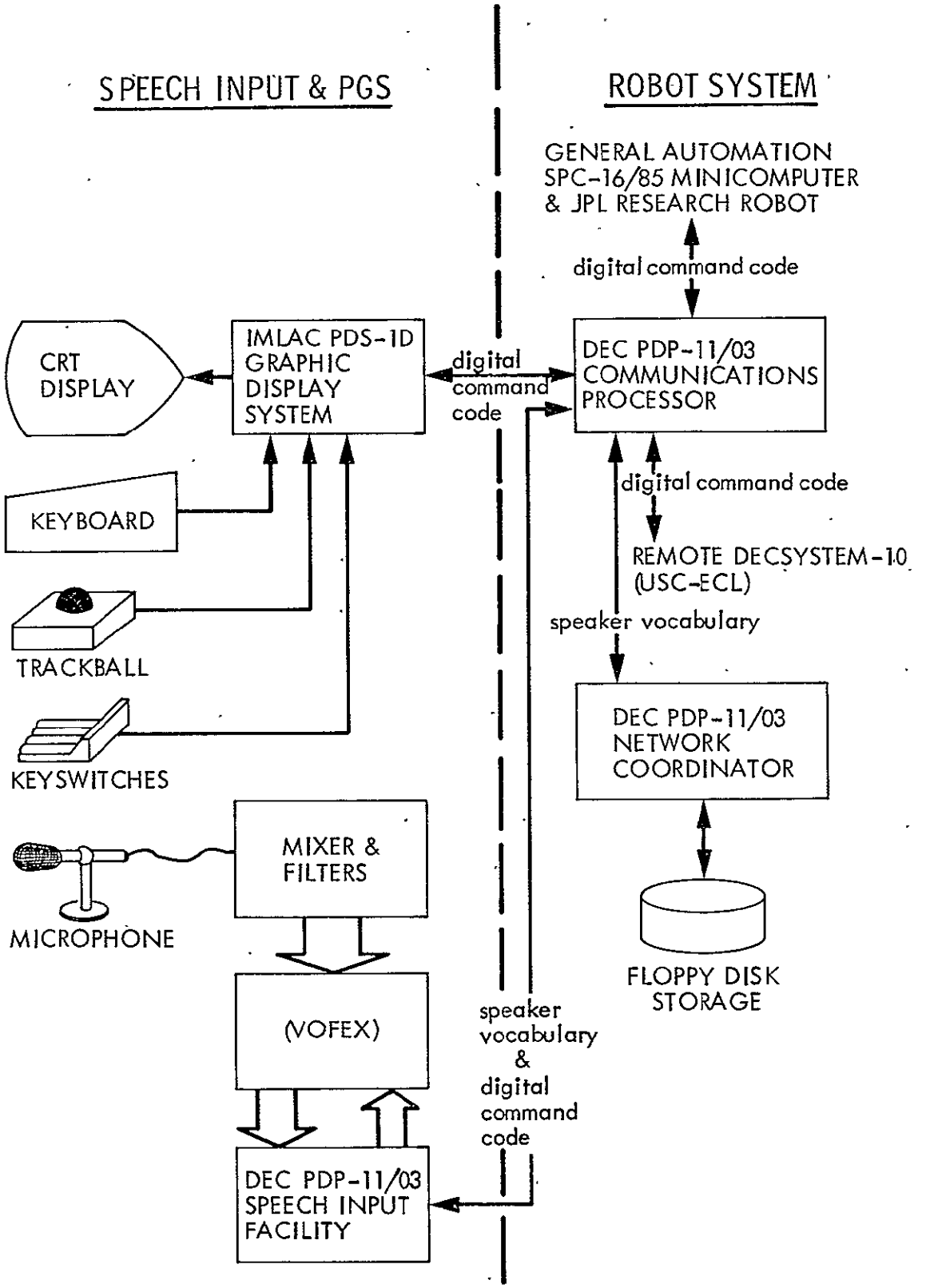


figure 3.5.1 Speech Input Facility - Robot System Interface

The initial utilization of the recognition process was to add voice input to the prototype ground subsystem (PGS) of the robotics research program. PGS is used as a control node for the robot system and is responsible for the graphic displays of subsystem states. In this initial application, the digital codes representing voice commands are forwarded to PGS by the communications subsystem and are processed by PGS as are its other input channels: keyboard, trackball and keyswitches.

To maintain flexibility of input form in using the PGS subsystem, the user can also specify commands via the PGS keyboard (i.e. choose not to use voice input). In this mode, the PGS subsystem forwards the ASCII input characters to the RECOGNIZEr process. The speech input facility processes character input in a similar manner to that of audio input. When the start and end of a word is detected, (a carriage return character represents word termination), the system checks the user's vocabulary file (appendix D) for the given input command, and obtains the digital code for the command. The syntactic constraints are enforced by insuring that the digital code of the input matches one assigned to an available node given the current state of the recognizer. If successful in these operations, the digital command code is returned and the state is updated; otherwise, a "no-match" response is generated as occurs for

the audio input mode.

The speech input and output processes execute in the same LSI-11 microcomputer. The word recognition process has priority over the processor as a result of its real-time characteristics. For this reason, the speech input process at specific points during its recognition operation lends the LSI-11 processor for a limited time to the speech output process. The speech input system is interrupt driven and no loss of data results. The word recognizer continues to "listen" and to collect information from the VOFEX describing the next utterance while data compression, normalization, comparison and classification is executing, and also while the processor is temporarily assigned to the voice output process.

CHAPTER 4 - THE AUTOMATIC VOICE OUTPUT SYSTEM

Voice response is a tool to be considered and utilized where applicable for computer output in much the same manner as one would select a hard copy or a CRT terminal. People react more immediately to the human voice than to any other means of communication. People are keyed to respond quickly to the spoken word [DATA 74]. Speech output was chosen to help provide a flexible system of communicating global information between the computer and user and is used in parallel with the other output channels: IMLAC graphics, CRT text and printed output from the remote Decsystem 10.

4.1 General Description

The robot voice output system is used to automatically inform the user of a critical system state, or as the result of a query, to communicate to the user the current status of a subsystem execution. For example, if the path planning subsystem determined that there did not exist a path to the desired site along which the vehicle could maneuver, then a short message conveying this could be voiced. If the manipulator arm was commanded to place a rock in an experiment bin, and upon attempting to lift the sample found that it was too heavy for the arm mechanism, another message

could be voiced. As the result of a user's request that the current state of the integrated system operation be output, such phrases as "vision 3-D correlation proceeding" or "manipulator local sensing proceeding" could be voiced.

The following properties characterize the application and requirements of the J.P.L. voice output system:

- short phrase voicings
- voicings are fixed in content
- medium sized, extensible repertoire of voicings
- rapid response to voicing commands (minimum delay)
- understandability of voicings
- cooperative user environment
- must execute on a DEC PDP-11/Ø3 microcomputer
- flexible software design and interface

Two methods for producing voice output are generation by means of stored digitized speech and speech synthesis. Speech can be reproduced by digitizing the original sound, storing its representation and later using digital-to-analog conversion techniques to revoice it. One can store digitized speech in ROM or RAM and then clock it out at the proper rate, smoothing the output by a low pass filter. This procedure requires the use of large amounts of storage and therefore is very costly and can only accommodate small

vocabularies or a few short phrases.

Phrases are composed of words, and words are made up of phonemes. In general, regardless of the variety of written spellings of a word, there exists only one phonetic spelling (string of phonemes).

Synthetic speech is not as clear or distinct in its nature as is actual speech. Synthetic speech is usually achieved by stringing together the sounds generated for each phoneme comprising the word. The lack in clarity results largely from synthesizer transitions from phoneme to phoneme, and from improper phoneme segment durations. The subtle shadings of intonation inherent in human speech cannot conveniently be reproduced by machine at this time, (i.e. intonation cannot fully be codified) [DATA 74]. The occassional recognition difficulty encountered due to this clarity problem is alleviated as users become accustomed to the synthesizer, (especially in a cooperative user environment with relatively short output utterances).

In using a voice synthesis rather than voice reproduction by means of digitized speech, less memory is required for the storage of the representations of each phrase. Each word is stored as a series of phoneme codes, not as a time series of speech wave values. A microcomputer controlled speech output system involving a voice

synthesizer requires less processor time and is less dependent upon performing real-time operations than one which directs the actual timing of successive output speech wave amplitudes. In a voice synthesis system, a number of phonemes can be passed from the microcomputer storage to the synthesizer buffer to be voiced depending upon the internal pacing of the synthesizing unit.

The speech output facility uses a VOTRAX VS-6.4 Audio Response System speech synthesizer [VOTR ØØ]. It is connected to a DEC PDP-11/Ø3 microcomputer by means of a serial interface. The same microcomputer is used for both the speech input and the speech output facilities.

The VOTRAX system utilizes a ROM storage unit which contains 63 phoneme sounds comprising the Standard American English dialect. There are only 38 distinct phonemes in the set, as 25 of the sounds are actually different length voicings of the principals. Other characteristics of the VOTRAX system include an input buffer to accommodate the difference between the data rate of the phonemes input and the rate in which they are used by the synthesizer to produce the pacing of the sounds output, and a modification mechanism to alter the production of phonemes based upon their immediate phonemic context. Four levels of inflection can be applied to each phoneme:

4.2 Organization and Operation

The software comprising the speech output facility is responsible for the production of a specific phrase upon request from a robot subsystem, (e.g. vision, arm). These utterances are static in content but extensible in number. Sample output utterances are listed in figure 4.2.1.

```
"laser generating environment map"
"rover encountering steep terrain"
"vision reports no objects detected in scene"
"scene analysis completed, select object of interest"
"arm unable to reach object"
"object tracking active for rover repositioning"
"arm unable to grasp object, object too large"
"arm unable to move object, object too heavy"
"load new speaker vocabulary"
```

figure 4.2.1
Sample Output Utterances

In the selection of an utterance request format, several choices were possible. The actual word could be used to represent the request. The phonetic description (with inflections) expressed as an ASCII string could be utilized. The VOTRAX command code to which the ASCII phoneme string must be translated could be used. And finally, a digital code could be used to represent the word. For example, for the word "communicate" to be voiced, the following codes could be used:

- "communicate"
  (the actual word, character by character)

- "2K 1UH2 2M 1Y1 1U1 1N 1N 1I1 1K 1A1 1AY 1Y1 1T"
  (the VOTRAX phonemes, with inflections, expressed as an ASCII string)

- 131 061 114 042 067 015 015 013 031 006 041 042 052
  (the VOTRAX instruction codes, expressed as 8-bit octal bytes)

- 117
  (a digital code assigned to the word)

For each of these choices, a tradeoff is made between the speech output facility processing responsibility and that of the requesting subsystem and communications link. For example, if the VOTRAX instruction code were to be sent by the subsystem, the speech output handler would pass the received code to the VOTRAX voice synthesizer, and the subsystem would be responsible for the storage, retrieval and transmitting of the substantial data volume representing the voicing. If a digital code were to be used, the subsystem would transmit to the speech output facility a single value representing the word or utterance to be voiced, and the voice output processor would be required to translate the code into the desired VOTRAX instructions by means of a code file and translation tables.

The output utterances require extended storage (e.g. disk) and must be expressed in a form which will allow for easy modification of utterance content as well as for the alteration of phonetic description and inflection assignment. The most convenient form in which to represent

words is by phonetic composition. Programs exist for translating text to phonemes [ELOV 76]; dictionaries are available for providing the phonetic spelling of words. For these reasons, a phonetic representation of words and phrases stored on the microcomputer network floppy disk was chosen.

The speech output facility interface to the robot system is illustrated in figure 4.2.2. Each output utterance(phrase) is assigned a digital code to be used by the robot system. For a subsystem to make a output request, it transmits to the communications LSI-11 microprocessor the utterance code, with the speech synthesizing process as its destination. The communications processor retrieves from the floppy disk the phonetic representation of the utterance and forwards it to the speech output facility. The voice output process then buffers up the entire message and translates it from its phonetic representation to the VOTRAX instruction form. These instructions are loaded into the speech synthesizer with the necessary controls signals to achieve the desired utterance. This system organization results in the rapid issuing of verbal responses by minimizing the volume of data which must be handled through the communications subsystem and extended storage.

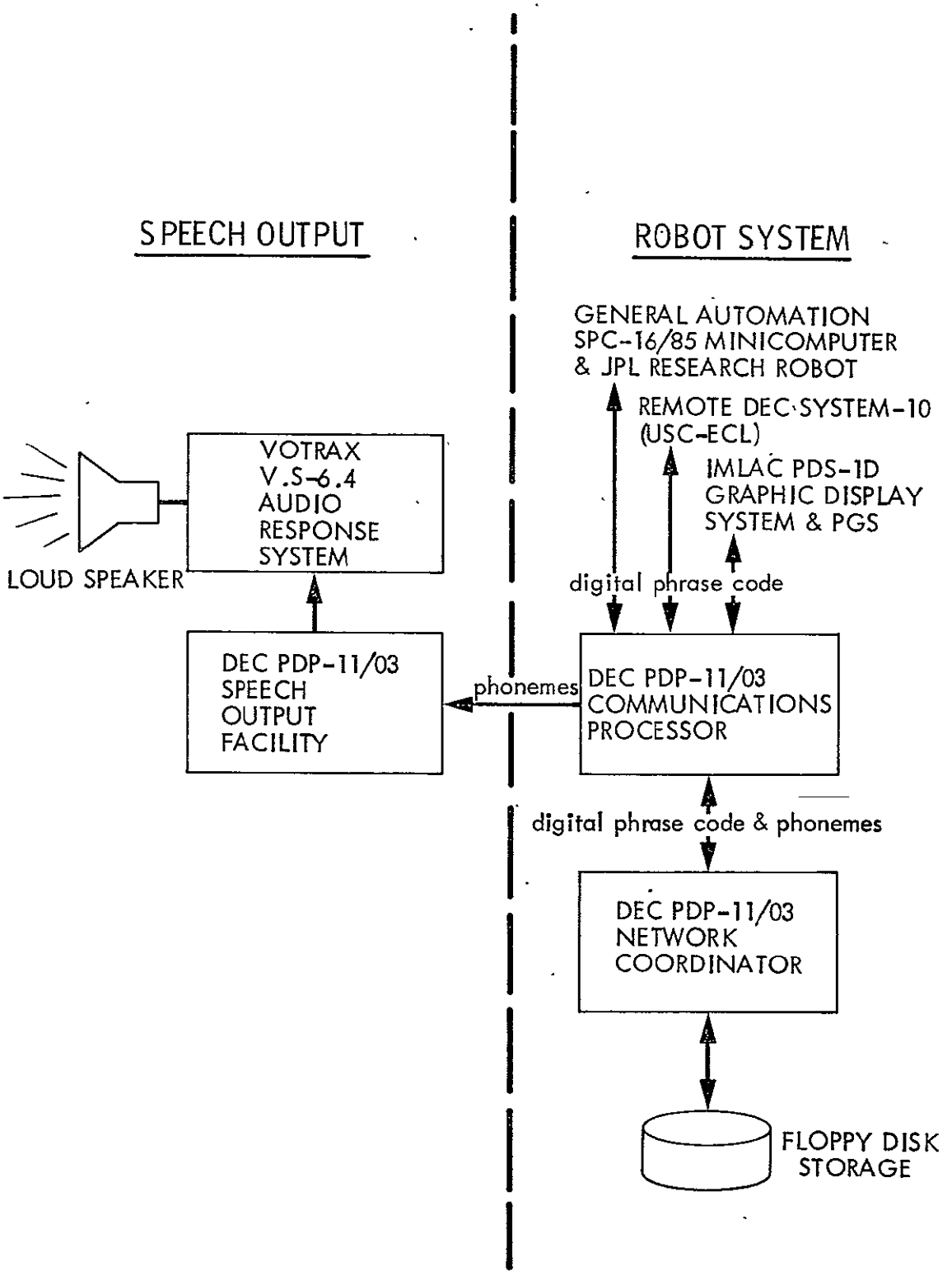


figure 4.2.2 Speech Output Facility - Robot System Interface

As noted in section 3.5, the speech output and voice input processes execute in the same LSI-11 microcomputer. The voice output process has a lower priority than the word recognition process. The VOTRAX speech synthesizer has data buffering capabilities and is not as time dependent as the speech input process. The speech output process also does not require as much processor time as does the voice input process. The software for these processes was designed separately permitting their integration into the robot system as individual facilities (subsystems).

CHAPTER 5 - CONCLUSION

Given the specific control application and the hardware constraints, speech input and output facilities were implemented into the J.P.L. robot system. Voice commands from an extensible vocabulary provide a user convenient input channel to question, direct and simulate the performance of the robot system and individual subsystems. The speech synthesis process represents an additional output channel to be used in parallel with the hard copy units and CRT displays. These new facilities provide the J.P.L. system with an overall control capability which was previously desired but not available.

Problems were encountered and dealt with in both individual speech input and output designs. In developing a word recognition system, the requirements with regards to vocabulary size, processing environment and cost, and the operational constraints of accuracy rate and speed, were difficult to reconcile. In achieving the desired word recognition performance, fast and efficient compression, normalization, comparison and classification algorithms had to designed and then implemented as PDP-11 assembly language routines. The PDP-11/03 microcomputer has a limited instruction set and slow processing speed. A hardware

feature extractor (VOFEX) was required to process the data volume necessary for isolated word recognition. The VOFEX was designed and built based upon the requirements of the speech input processor. The close to real-time execution condition necessitated the use of computationally simple assembly routines suitable for the isolated word robotic application. Syntactic constraints were incorporated into the vocabulary to improve recognition accuracy and speed.

The most severe problem encountered in the speech input work arose from the non-ideal nature of the filters used to separate the fundamental frequencies of speech. This problem was dealt with, (see section 3.2), by making adjustments upon the bandpass filters and the VOFEX hardware.

In the operation of the speech output facility, data communication load characteristics and phrase storage requirements could place heavy demands upon the LSI-11 microprocessor and the J.P.L. robot system. Through coding techniques and choice of subsystem communication protocol, the voice output facility was integrated into the remainder of the robot system and is able to execute along with the speech input process in the same microcomputer.

APPENDIX A
VOice Feature EXtractor (VOFEX)

The Voice Feature Extraction hardware is responsible for the gathering of zero-crossing and energy information for each of the four frequency bandpasses. Zero-crossing counts and measures proportional to average energies are accumulated over a period of time ("window") as dictated by the recognition software. The interface between the VOFEX and the recognizer consists of a DRV-11 parallel interface unit and an ADAC Corporation Model 600-LSI-11 Data Acquisition and Control System; both boards reside in the PDP-11/03 microcomputer.

Each of the four analog CROWN bandpass filter outputs are applied to separate sets of zero-crossing and energy circuits. The four circuit groups are identical except for the amplification factor necessary to scale the inputs to the -10 to +10 voltage range. Comparators are used to detect zero-crossings; the digital outputs of the comparators are applied to pairs of four-bit counters to accumulate the axis-crossing counts.

The zero-crossing counts for the four bands are routed to a selection module. The recognizer software selects from which band, the zero-crossing value (eight bits) will be

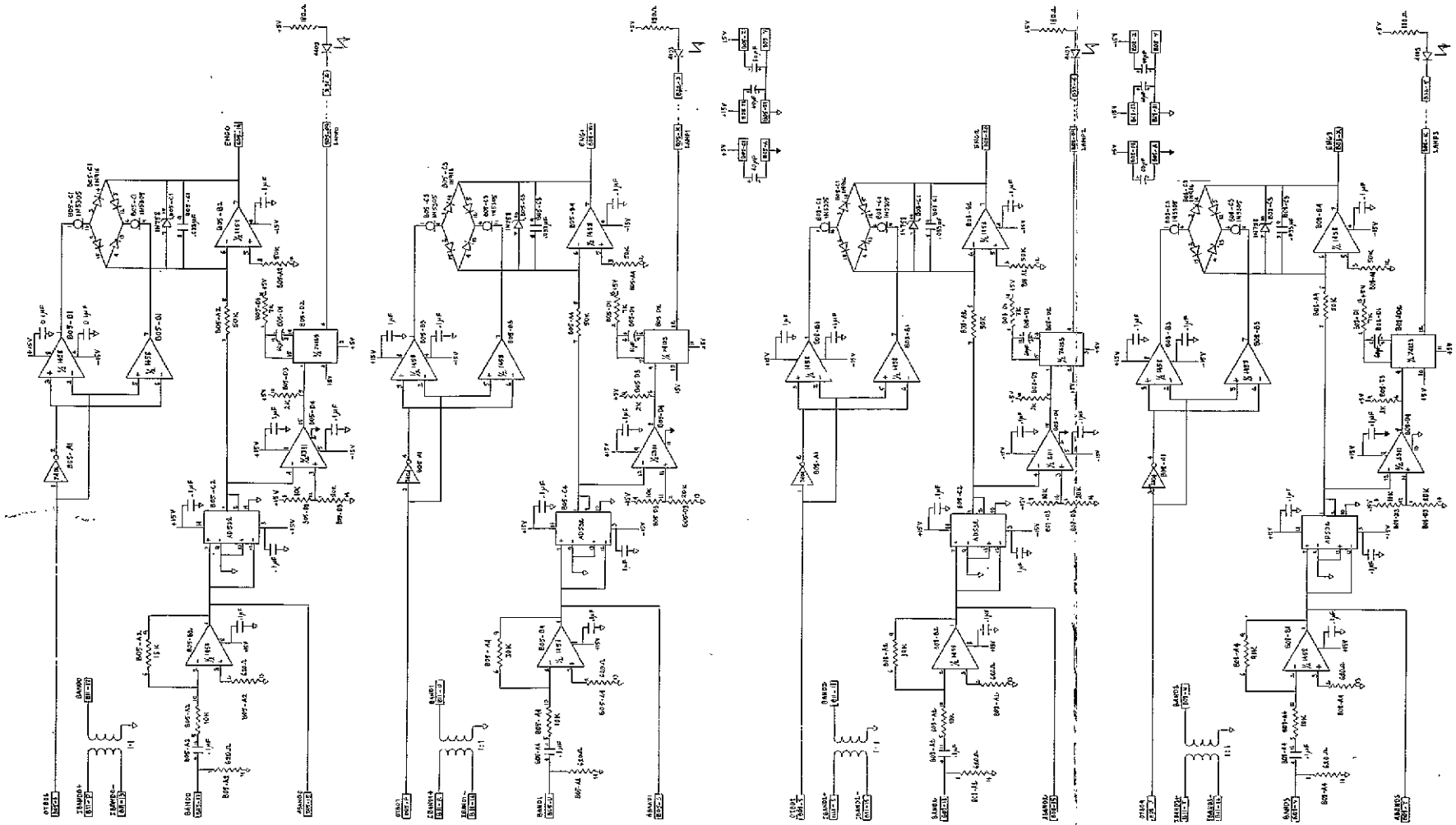
applied to the parallel interface input bus. This is accomplished by placing the appropriate two-bit code in the interface output register. Four additional output register bits are used to individually place the counters in a cleared or counting mode.

Average energy measures are produced through analog means. The amplified inputs are squared and scaled to obtain amplitudes in the Ø to +1Ø voltage range for normally voiced speech. Speaker inputs which saturate this VOFEX amplitude range are clipped and trigger LEDs as a warning indication. The amplitudes are then summed through use of an integrating capacitor circuit. Capacitor voltages are provided as inputs to the ADAC analog-to-digital converter and can be sampled at any time by the recognizer. Four parallel output register bits (separate from the six previously specified) are used to individually place the integrating capacitors in either a cleared or summing mode.

Schematics of the Voice Feature Extraction hardware are provided on the following pages.

FOLDOUT FRAME 1

FOLDOUT FRAME 2



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FOLDOUT FRAME 1

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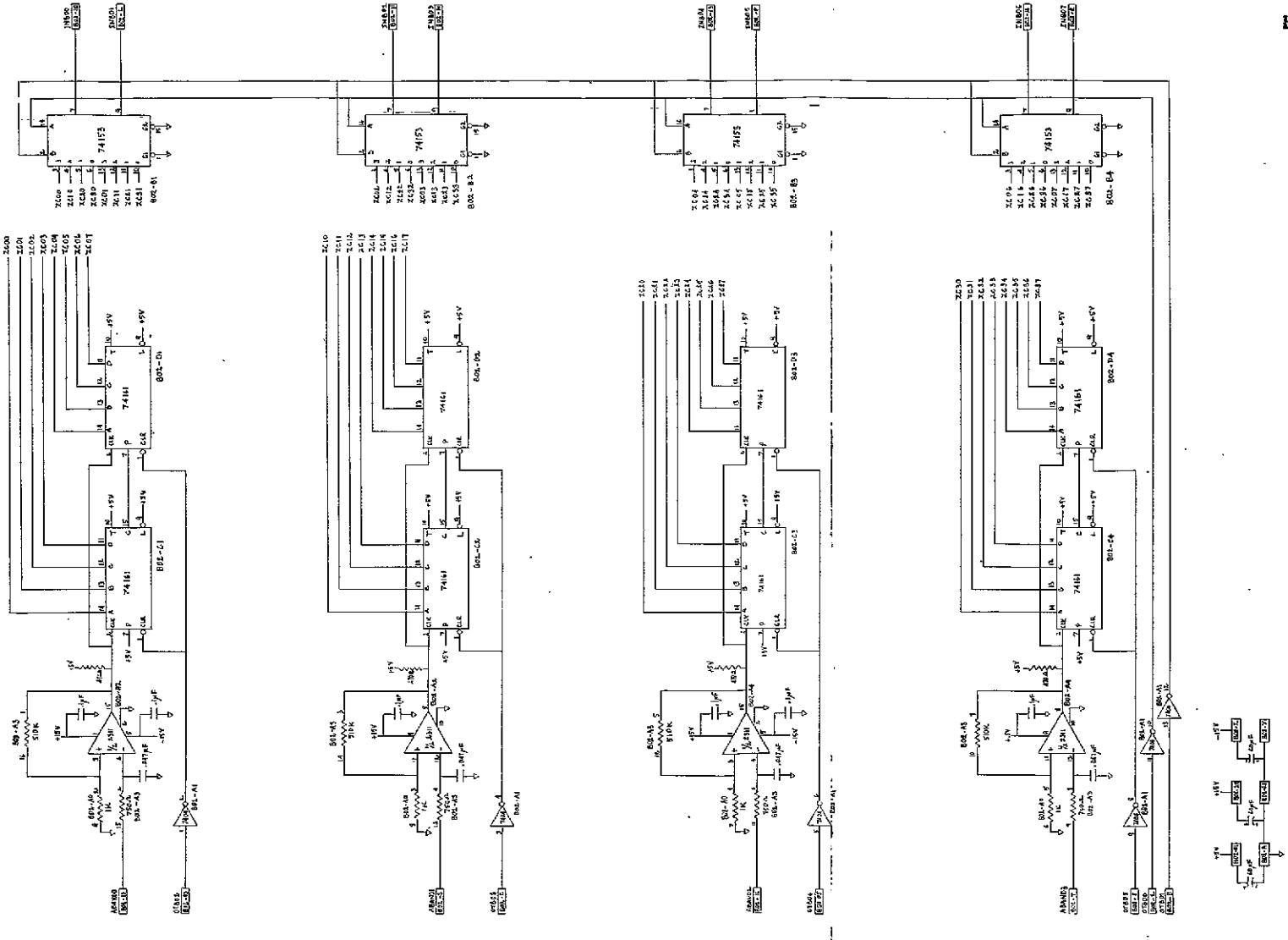

APPENDIX B
Recognition System Parameters

MAXIMUM VOCABULARY SIZE: 100 commands **

NUMBER OF FREQUENCY BANDS: 4

BAND SETTINGS:

band #	frequency range (Hz.)
0	250 - 450
1	700 - 1400
2	1850 - 2500
3	3000 - 4000

WINDOW PERIOD: 10 milliseconds **

MAXIMUM LENGTH OF COMMAND: 3 seconds **

SILENCE DURATION REQUIRED TO TRIGGER END-UTTERANCE DETECT:
150 milliseconds (15 window periods) *
MINIMUM UTTERANCE DURATION TO BE CONSIDERED VALID INPUT:
150 milliseconds (15 window periods) *

LENGTH OF NORMALIZED Z/C BAND VECTOR: 16 segments **
LENGTH OF NORMALIZED ENERGY BAND VECTOR: 16 segments **

NORMALIZED Z/C BAND VECTOR STORAGE: 16 bytes (8 words) **
NORMALIZED ENERGY BAND VECTOR STORAGE: 16 bytes (8 words) **

PROTOTYPE STORAGE SIZE: 128 bytes (64 words) per command **

(*) - parameter can be changed by loading new vocabulary file.
(**) - parameter can be changed by reassembling source code.

APPENDIX C
Robotic Vocabulary Description

This appendix is intended to supply additional information regarding the data structures produced by the vocabulary generation program and used by the learning and recognition routines. The user first defines the vocabulary in a hierachical manner, providing node levels and digital codes for each command word or phrase, (values are in octal form). A sample robotic application vocabulary description appears below:

```
1  ROOT       30,
   2  SUMMARY    31,
      3  STATUS     32,
      3  DISPLAY    33,
         4  UPDATE     34,
         4  SIMULATE   35,
         4  FREEZE     36,
   2  ARM        37,
      3  STATUS     32,
      3  DISPLAY    33,
         4  UPDATE     34,
         4  SIMULATE   35,
         4  FREEZE     36,
      3  JOINTS     40,
         4  UPDATE     34,
         4  SIMULATE   35,
         4  FREEZE     36,
      3  TORQUE     41,
         4  UPDATE     34,
         4  SIMULATE   35,
         4  FREEZE     36,
      3  WEIGHT     42,
      3  SENSE      43,
         4  UPDATE     34,
         4  SIMULATE   35,
         4  FREEZE     36,
   2  EYE        44,
      3  STATUS     32,
      3  DISPLAY    33,
         4  UPDATE     34,
         4  SIMULATE   35,
         4  FREEZE     36,
      3  VISION     45,
         4  AUTOMATIC  46,
         4  SEGMENT    47,
         4  GROW       50,
         4  LOCATE     51,
         4  MAP        52,
      3  CAMERAS    53,
         4  FOCUS      54,
            5  UPDATE     34,
            5  SIMULATE   35,
            5  FREEZE     36,
         4  CONTRAST   55,
            5  UPDATE     34,
            5  SIMULATE   35,
            5  FREEZE     36,
         4  TRACK      56,
            5  UPDATE     34,
            5  SIMULATE   35,
            5  FREEZE     36,
   2  ROVER      57,
```

```
      3  STATUS     32,
      3  TERRAIN    60,
      3  PATH       61,
      3  GYRO       62,
         4  UPDATE     34,
         4  SIMULATE   35,
         4  FREEZE     36,
      3  NAVIGATE   63,
         4  UPDATE     34,
         4  SIMULATE   35,
         4  FREEZE     36,
   2  GLOBAL     0,
      3  SPEAKER    1,
      3  OFF        2,
      3  ON         3,
      3  QUIT       4 ;
```

The following illustration represents the above vocabulary description in its tree format:

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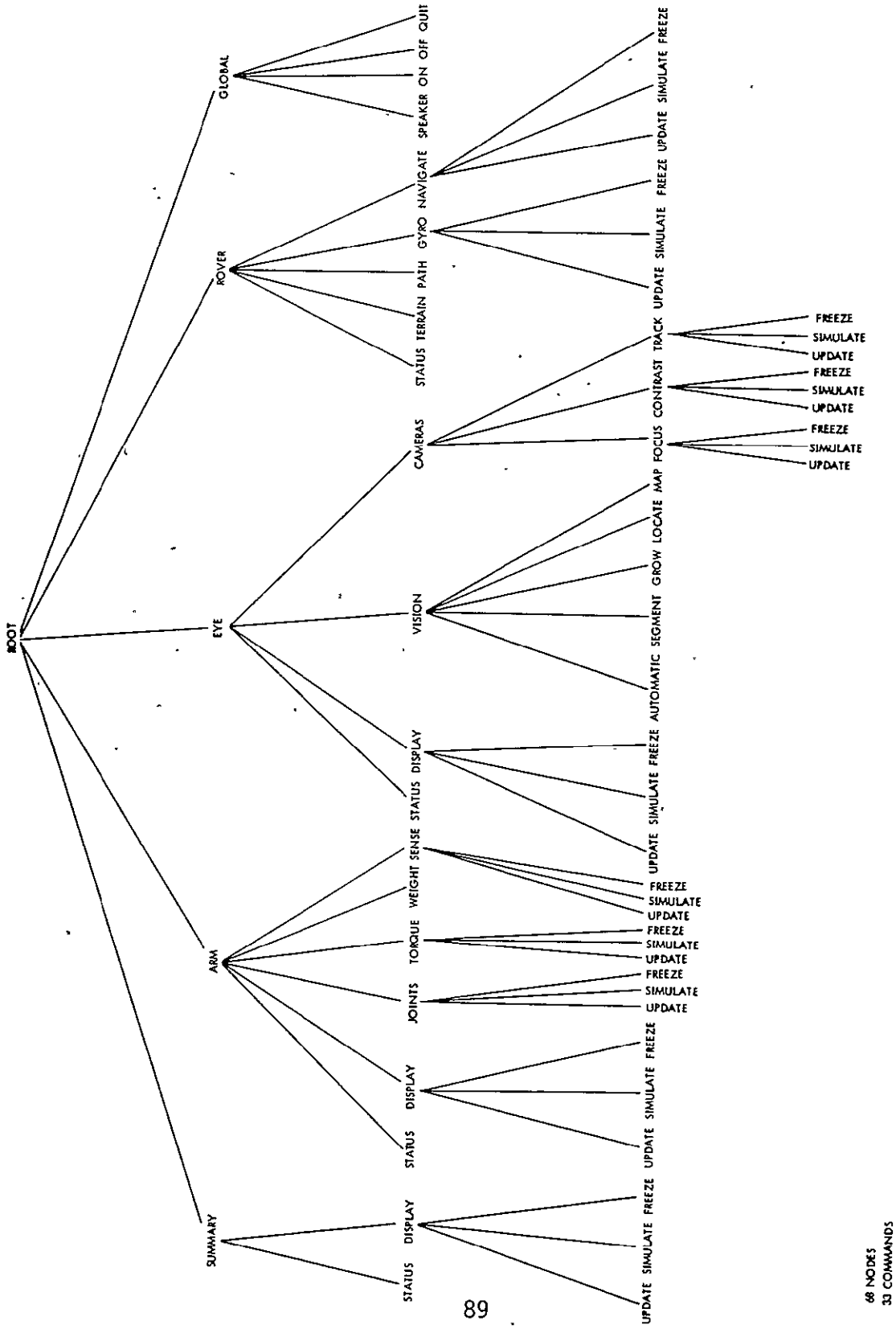

The vocabulary generation program receives as its input the hierarchical description of the vocabulary, and produces an untrained vocabulary description file consisting of speaker dependent variables (see appendix D), syntactic constraint rules, command prototype storage and a digital code/command entry table. The command prototype storage area remains vacant until the user trains the recognizer for the given vocabulary by means of the LEARN program. The following is the execution summary produced by the VOCGEN program for the sample vocabulary. The LLSCTAB offset represents the relative address in the syntactic constraint structure for the given node (not digital command) entry. The syntactic constraint structure lists for each node in the vocabulary tree, the relative address in the prototype storage for the normalized command data, the digital command code and the LLSCTAB offsets for the command nodes which can legally follow. ACGLOBAL is the LLSCTAB offset for the GLOBAL command subtree.

*** VOCABULARY GENERATION EXECUTION SUMMARY ***

VOCABULARY STRUCTURE TABLE:

LEVEL#	WORD ENTRY	DIG. CODE	LLSCTAB OFFSET
000001	ROOT	000030	000000
000002	SUMMARY	000031	000020
000003	STATUS	000032	000044
000003	DISPLAY	000033	000052
000004	UPDATE	000034	000076
000004	SIMULATE	000035	000104
000004	FREEZE	000036	000112
000002	ARM	000037	000120
000003	STATUS	000032	000154
000003	DISPLAY	000033	000162
000004	UPDATE	000034	000216
000004	SIMULATE	000035	000224
000004	FREEZE	000036	000232
000003	JOINTS	000040	000240
000004	UPDATE	000034	000274
000004	SIMULATE	000035	000302
000004	FREEZE	000036	000310
000003	TORQUE	000041	000316
000004	UPDATE	000034	000352
000004	SIMULATE	000035	000360
000004	FREEZE	000036	000366
000003	WEIGHT	000042	000374
000003	SENSE	000043	000402
000004	UPDATE	000034	000436
000004	SIMULATE	000035	000444
000004	FREEZE	000036	000452
000002	EYE	000044	000460
000003	STATUS	000032	000510
000003	DISPLAY	000033	000516
000004	UPDATE	000034	000546
000004	SIMULATE	000035	000554
000004	FREEZE	000036	000562
000003	VISION	000045	000570
000004	AUTOMATIC	000046	000624
000004	SEGMENT	000047	000632
000004	GROW	000050	000640
000004	LOCATE	000051	000646
000004	MAP	000052	000654
000003	CAMERAS	000053	000662
000004	FOCUS	000054	000712
000005	UPDATE	000034	000740
000005	SIMULATE	000035	000746
000005	FREEZE	000036	000754
000004	CONTRAST	000055	000762
000005	UPDATE	000034	001010

000005	SIMULATE	000035	001016
000005	FREEZE	000036	001024
000004	TRACK	000056	001032
000005	UPDATE	000034	001060
000005	SIMULATE	000035	001066
000005	FREEZE	000036	001074
000002	ROVER	000057	001102
000003	STATUS	000032	001134
000003	TERRAIN	000060	001142
000003	PATH	000061	001150
000003	GYRO	000062	001156
000004	UPDATE	000034	001210
000004	SIMULATE	000035	001216
000004	FREEZE	000036	001224
000003	NAVIGATE	000063	001232
000004	UPDATE	000034	001264
000004	SIMULATE	000035	001272
000004	FREEZE	000036	001300
000002	GLOBAL	000000	001306
000003	SPEAKER	000001	001326
000003	OFF	000002	001334
000003	ON	000003	001342
000003	QUIT	000004	001350
177777		177777	001356

DIGITAL CODE- WORD ENTRY TABLE:

DIG. CODE	WORD ENTRY
000030	ROOT
000031	SUMMARY
000032	STATUS
000033	DISPLAY
000034	UPDATE
000035	SIMULATE
000036	FREEZE
000037	ARM
000040	JOINTS
000041	TORQUE
000042	WEIGHT
000043	SENSE
000044	EYE
000045	VISION
000046	AUTOMATIC
000047	SEGMENT
000050	GROW
000051	LOCATE
000052	MAP
000053	CAMERAS
000054	FOCUS
000055	CONTRAST
000056	TRACK
000057	ROVER
000060	TERRAIN
000061	PATH
000062	GYRO
000063	NAVIGATE
000000	GLOBAL
000001	SPEAKER
000002	OFF
000003	ON
000004	QUIT

NDWDS: 000041

SYNTACTIC CONSTRAINT STRUCTURE:

```
[LLSCTAB OFFSET]:      [PROTOTYPE OFFSET]      [DIG. CODE]
>[LLSCTAB OFFSET OF LEGAL COMMAND1]
>[LLSCTAB OFFSET OF LEGAL COMMAND2]
>[LLSCTAB OFFSET OF LEGAL COMMAND3]
> ETC.

000000:         000000          000030
>000000
>000020
>000120
>000460
>001102

000020:         000200          000031
>000020
>000044
>000052
>000120
>000460
>001102
>000000

000044:         000400          000032

000052:         000600          000033
>000052
>000076
>000104
>000112
>000044
>000020
>000000

000076:         001000          000034

000104:         001200          000035

000112:         001400          000036

000120:         001600          000037
>000120
>000154
>000162
>000240
>000316
>000374
>000402
>000460
>001102
>000020
```

```
>000000

000154:         000400          000032

000162:         000600          000033
>000162
>000216
>000224
>000232
>000240
>000316
>000374
>000402
>000154
>000120
>000000

000216:         001000          000034

000224:         001200          000035

000232:         001400          000036

000240:         002000          000040
>000240
>000274
>000302
>000310
>000316
>000374
>000402
>000162
>000154
>000120
>000000

000274:         001000          000034

000302:         001200          000035

000310:         001400          000036

000316:         002200          000041
>000316
>000352
>000360
>000366
>000374
>000402
>000240
>000162
>000154
```

```
>000120
>000000

000352:         001000          000034

000360:         001200          000035

000366:         001400          000036

000374:         002400          000042

000402:         002600          000043
>000402
>000436
>000444
>000452
>000374
>000316
>000240
>000162
>000154
>000120
>000000

000436:         001000          000034

000444:         001200          000035

000452:         001400          000036

000460:         003000          000044
>000460
>000510
>000516
>000570
>000662
>001102
>000120
>000020
>000000

000510:         000400          000032

000516:         000600          000033
>000516
>000546
>000554
>000562
>000570
>000662
>000510
>000460
```

```
>000000

000546:         001000          000034

000554:         001200          000035

000562:         001400          000036

000570:         003200          000045
>000570
>000624
>000632
>000640
>000646
>000654
>000662
>000516
>000510
>000460
>000000

000624:         003400          000046

000632:         003600          000047

000640:         004000          000050

000646:         004200          000051

000654:         004400          000052

000662:         004600          000053
>000662
>000712
>000762
>001032
>000570
>000516
>000510
>000460
>000000

000712:         005000          000054
>000712
>000740
>000746
>000754
>000762
>001032
>000662
>000000
```

```
000740:         001000          000034

000746:         001200          000035

000754:         001400          000036

000762:         005200          000055
>000762
>001010
>001016
>001024
>001032
>000712
>000662
>000000

001010:         001000          000034

001016:         001200          000035

001024:         001400          000036

001032:         005400          000056
>001032
>001060
>001066
>001074
>000762
>000712
>000662
>000000

001060:         001000          000034

001066:         001200          000035

001074:         001400          000036

001102:         005600          000057
>001102
>001134
>001142
>001150
>001156
>001232
>000460
>000120
>000020
>000000

001134:         000400          000032
```

```
001142:         006000          000060

001150:         006200          000061

001156:         006400          000062
>001156
>001210
>001216
>001224
>001232
>001150
>001142
>001134
>001102
>000000

001210:         001000          000034

001216:         001200          000035

001224:         001400          000036

001232:         006600          000063
>001232
>001264
>001272
>001300
>001156
>001150
>001142
>001134
>001102
>000000
001264:         001000          000034

001272:         001200          000035

001300:         001400          000036

001306:         007000          000000
>001306
>001326
>001334
>001342
>001350

001326:         007200          000001

001334:         007400          000002

001342:         007600          000003
```

```
001350:         010000          000004

ACGLOBAL: 001306

 VOCABULARY GENERATION SUCCESSFUL-
```

APPENDIX D
User Vocabulary File

A user's vocabulary file is composed of four sections: the speaker dependent variables, the syntactic constraint rules, the command prototypes and the digital code/command entry table. The speaker dependent variable section contains the parameters used in the start and end detect of an utterance, the vector difference weights used by the classification routines, the thresholds used by the decision procedure and the digital codes of special global commands needed by the recognition supervisor.

The syntactic constraint area holds the tree-structured vocabulary information and is organized in a preorder fashion. For each command node in the tree, its prototype offset address and digital code of the entry is stored, along with a lists of valid(accessible) nodes available from the given state.

The prototype storage section holds the normalized zero-crossing and energy information for each distinct command (digital code) in the vocabulary. Given the current normalization techniques used, a vocabulary of 100 commands would require 12.5K bytes for prototype storage; (56K bytes of storage are available in the DEC PDP-11/03

microcomputer).

The digital code/command entry table stores the actual command identity (in ASCII character format) for each digital code. This table is used by the recognizer program to process keyboard input and by the learning program to prompt the user during vocabulary training.

A sample user vocabulary file follows (values are in octal form). Sample syntactic constraint data structure and digital code/command entry table can be found in appendix C.

```
SPEAKER DEPENDENT VARIABLES
ESTART:          ; MINIMUM ENERGIES NEEDED TO TRIGGER START
 30              ; BAND 0
 34              ; BAND 1
 24              ; BAND 2
 24              ; BAND 3

EEND:            ; MAXIMUM ENERGIES NEEDED TO TRIGGER END
 24              ; BAND 0
 30              ; BAND 1
 20              ; BAND 2
 20              ; BAND 3

ENDWCNT: 17      ; NUMBER OF CONSECUTIVE SILENCE WINDOWS
                 ; REQUIRED TO TRIGGER END DETECT
TOOSHORT: 17     ; UTTERANCE HAS TO BE LONGER THAN THIS LENGTH

ERANGE:          ; MINIMUM ENERGY VALUE RANGES FOR AFTER
                 ; NORMALIZATION, ELSE IGNORE INPUT
 50              ; BAND 0
 50              ; BAND 1
 40              ; BAND 2
 30              ; BAND 3

DECWEIGHTS:      ; FEATURE DECISION WEIGHTS
 1               ; BAND 0 - Z/C
 6               ; BAND 1 - Z/C
 6               ; BAND 2 - Z/C
 4               ; BAND 3 - Z/C
 1               ; BAND 0 - ENERGY
 2               ; BAND 1 - ENERGY
 2               ; BAND 2 - ENERGY
 1               ; BNAD 3 - ENERGY

MAXDIF: 6000     ; MAXIMUM DISTANCE BETWEEN UNKNOWN INPUT AND
                 ; BEST PROTOTYPE FOR ACCEPTANCE (THRESHOLD)
MINQUO: 114      ; CONFIDENCE RATIO x 64 MUST BE GREATER THAN
                 ; THIS VALUE (THRESHOLD)

MAXGCODE: 20     ; MAXIMUM GLOBAL COMMAND CODE
ILLEGAL: -1      ; DIGITAL CODE OF A NO-MATCH ENTRY
DCGLOBAL: 0      ; DIGITAL CODE OF "GLOBAL"
DCSPEAKER: 1     ; DIGITAL CODE OF "SPEAKER"
DCON: 2          ; DIGITAL CODE OF "ON"
DCOFF: 3         ; DIGITAL CODE OF "OFF"
DCEXIT: 4        ; DIGITAL CODE OF "EXIT"
```

```
SYNTACTIC CONSTRAINT STORAGE
ACGLOBAL: 1306 ; ABSOLUTE OFFSET IN LLSCTAB FOR "GLOBAL"
ENTRY
LLSCTAB:       ; STORAGE FOR 100 UNIQUE COMMANDS


PROTOTYPE PATTERN STORAGE
PROTOS:        ; STORAGE FOR 100 NORMALIZED COMMANDS


DIGITAL CODE/COMMAND ENTRY TABLE
NENTRY: 0      ; NUMBER OF UNIQUE COMMANDS IN VOCABULARY
ENTRYS:        ; A DIGITAL CODE/COMMAND SPELLING ENTRY FOR
               ; EACH COMMAND IN THE VOCABULARY
```

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