A TUTORIAL ON DIGITAL MEASUREMENT AND

ANALYSIS OF ANALOG VOLTAGE SIGNALS

by

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List of Figures

In this paper will be presented a discussion of digital measurement and analysis of analog voltage signals. Figure 1 is an example of the type of voltage signal to be discussed here. This discussion is based on the considerations necessary to accurately measure, using a digital conversion method, a continuous analog signal. Figure 2 shows a typical hardware configuration of a voltage measurement system which will be discussed in more detail in Appendix I. Part I contains hardware considerations for design of an analog voltage measurement system. Considerations for digitizing such as signal filtering, signal sampling, and choice of analog to digital converter will be discussed in terms of minimum requirements for accurate signal recovery. Considerations for analysis such as choice of main digital processor and choice of peripheral storage devices will be discussed in terms of minimum requirements for accurate signal processing. Part II contains software considerations for design of an analog voltage measurement system. Considerations such as data buffering and storing and signal amplitude and time scale recovery will be discussed in terms of accuracy of signal recovery as well as versatility of the software system. Some of the types of software available for signal analysis will be discussed. Illustrations are included wherever possible to help clarify this discussion. Appendix I contains the parameters for an electroencephalograph digital measurement system currently in use by Dr. Howard Shevrin, Senior Psychologist at The Menninger Foundation, Topeka, Kansas, which the author implemented on a Digital Equipment Corporation PDP-12.

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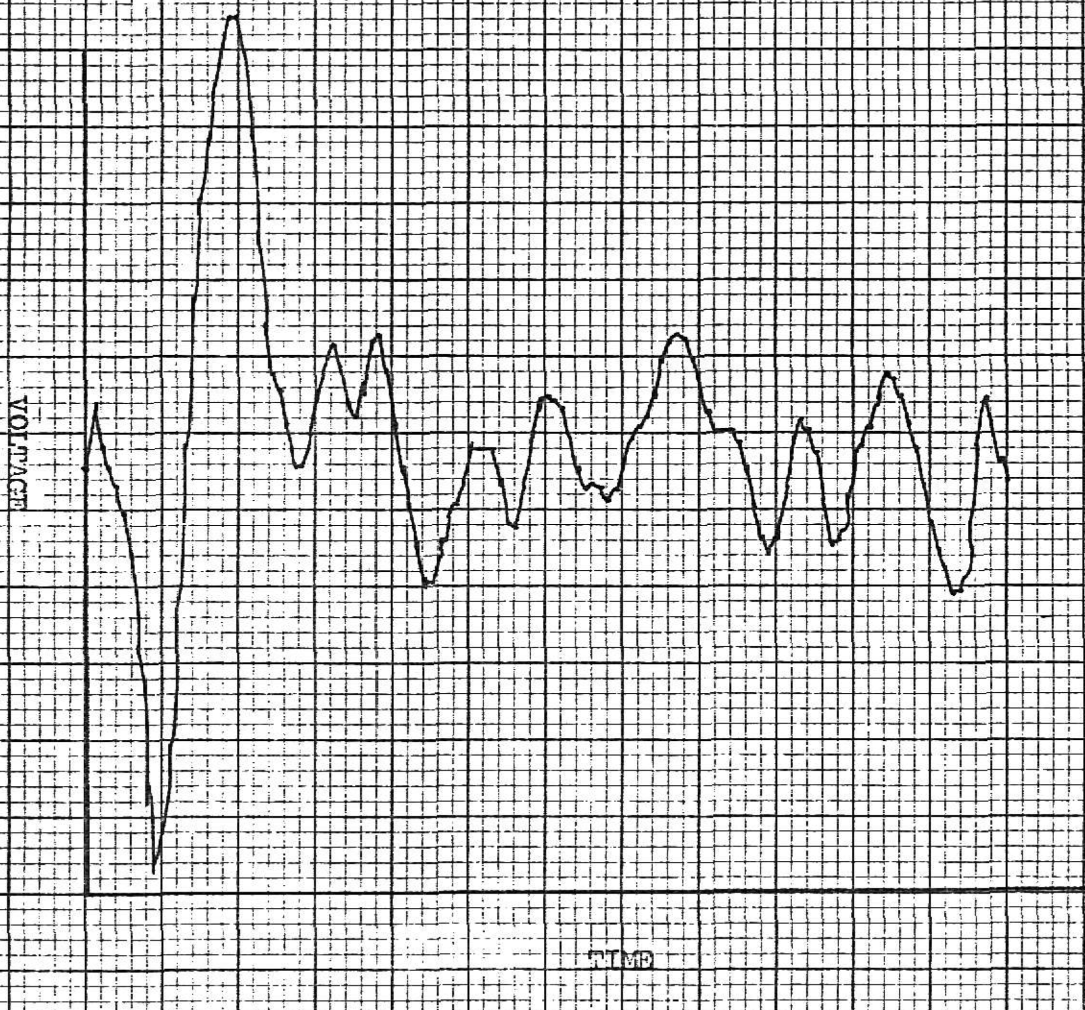
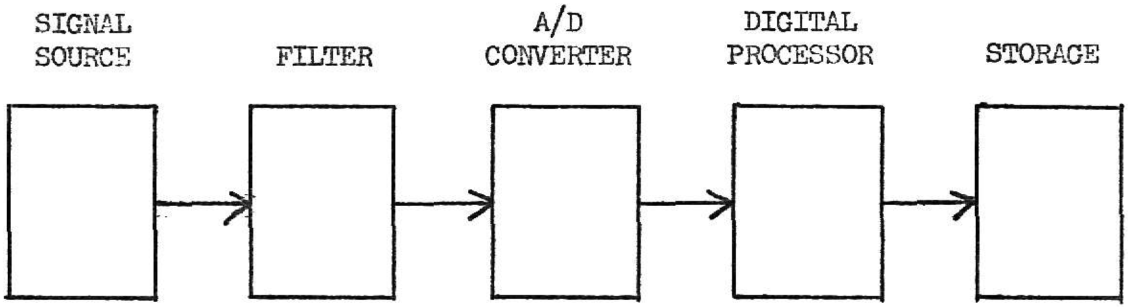


EXAMPLE OF ANALOG VOLTAGE SIGNAL

FIGURE 1



SAMPLE HARDWARE CONFIGURATION--VOLTAGE DIGITAL MEASUREMENT SYSTEM

FIGURE 2

PART I

HARDWARE CONSIDERATIONS FOR DESIGN OF AN ANALOG VOLTAGE DIGITAL ANALYSIS SYSTEM

1. Signal Filtering

It can be shown and will be discussed in section 2 that no matter what sampling rate is used to recover a wanted signal, if there is high frequency noise present in the signal a low pass filter must be used. The unwanted frequencies must be eliminated, yet the signal must come through the filter without distortion. The cut off frequency and roll off rate of a filtering system must be chosen so as to eliminate the unwanted high frequency signal without distorting the wanted frequencies. It may also be necessary to eliminate a narrow band of frequencies (such as 60 hertz) with notch filters. To the extent that the filtering is imperfect the recovery of the signal will also be imperfect. A technical discussion of the design of such a filtering system will not be included here. The reader may refer to Appendix I for the parameters of such a filtering system.

2. Signal Sampling

The sampling rate determines the accuracy to which a voltage input signal can be reconstructed.

2.1 Sample Timing

Timing control of the analog to digital converter is necessary. The speed of the control and the type necessary are dependent on the specific signal measurement application.

To accurately analyze an analog input signal, the signal must be sampled at a constant rate. This timing can be accomplished by using the processor cycle as the basis for timing, but it is more accurately done by using an external real-time clock which can be set to overflow or cause an interrupt at specified time intervals.

2.2 Sampling Theorem

Sampling theory considers the problem: "How short must the sampling interval be to insure that the variable does not vary without the observor's knowledge?". By sampling theory the answer to this question is based on the frequency of the signal one wants to recover, that is, the "frequencies of interest". Two variables must be considered in choosing the sampling rate which will accurately measure an input signal with known frequencies. The sampling rate must be equal to, or higher than, twice the highest frequency of interest in a particular application. In addition to this, it must be assured that no frequency higher than the frequencies of interest are present in the signal when the sampling occurs.

2.3 Sampling Theorem Application

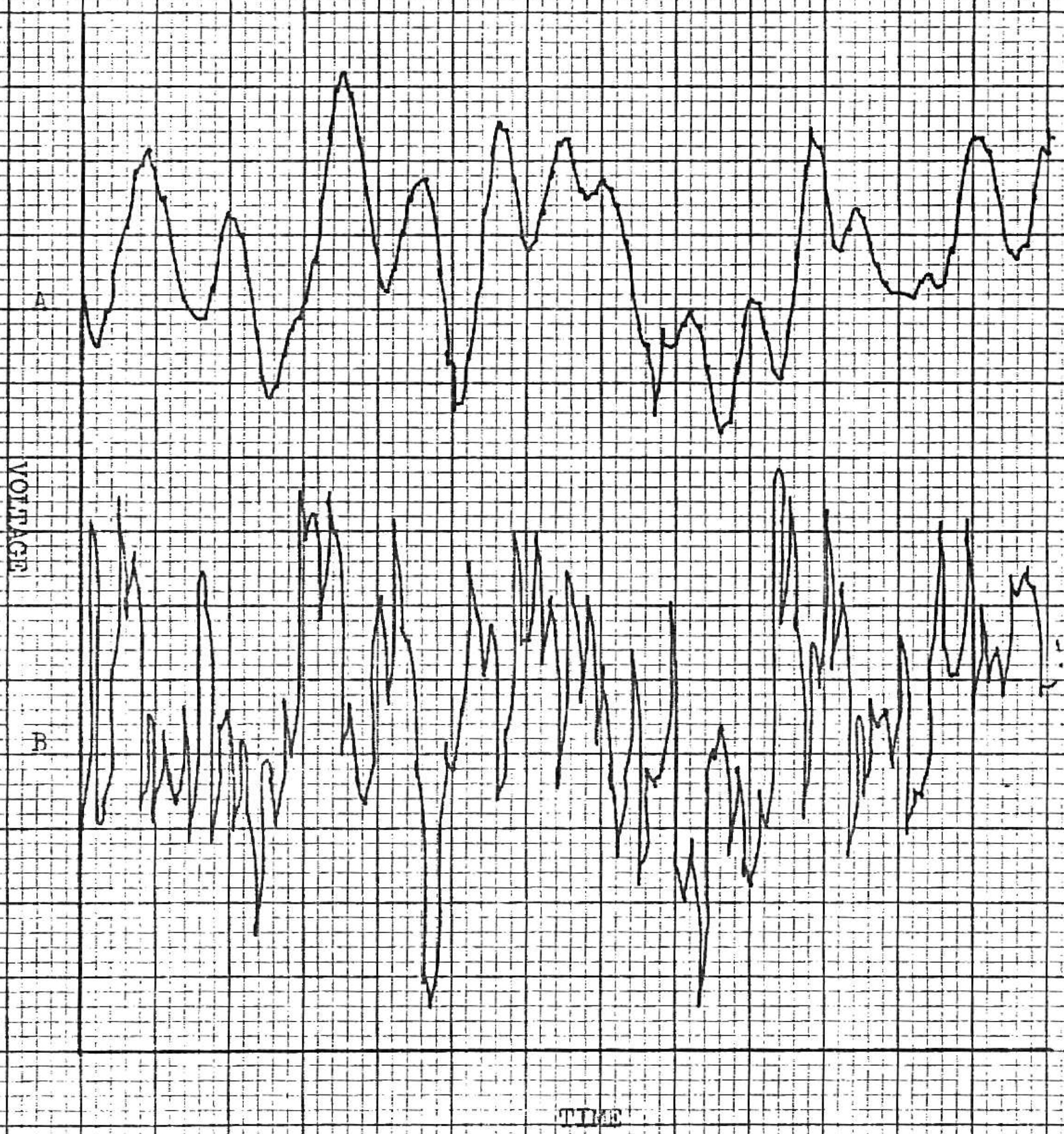
By the sampling theorem, choosing a sampling rate which will accurately recover a wanted input signal is more complex than simply sampling at a rate equal to, or higher than, twice the highest frequency of interest. If the high frequency noise components which may be present in a signal are not eliminated before sampling occurs, two difficulties in recovering the signal arise. If the sampling rate is chosen with the purpose of recovering only the frequencies of interest from the original signal (sampling rate is twice the highest frequency of interest) the presence of higher frequency components will cause "folding" of the frequency spectrum. That is, sampling superimposes the sample of components of higher frequency onto the component whose frequencies are less than the chosen frequencies of interest so that an unintelligible jumble results. If the sampling rate is chosen so that one may recover components which are much higher frequency than the frequencies of interest (sampling rate is greater than twice the highest frequency of interest) then the wanted frequencies will be obscured in the resulting sample by the "high frequency noise" which has been allowed into the sample. Figure 3 is an example of the latter case in which the sampling rate is high enough to recover a signal of much higher frequency than the frequencies of interest but the high frequency noise in the unfiltered data obscures the wanted frequencies. Since in either case the frequencies of interest are obscured or lost, a low pass filter must be used to eliminate the high frequency components prior to sampling. In practice, the sampling rate for a given application arrived at by the sampling theorem is the minimum sampling rate at which a signal can be recovered. The actual sampling rate one uses is determined by two main factors. Applications measuring input voltages involve the implicit measurement of time. If the measurement of time is important to a particular application the sampling rate determines the accuracy to which one can recover an occurrence latency. It would be ideal then to maximize the accuracy to which time can be recovered by maximizing the sampling rate. The sampling rate is limited by the necessity to store the digital values of the input signal. The amount of core storage as well as the amount of peripheral storage (tape or disc for example) available will determine the maximum feasible sampling rate. There are two other possible restrictions to the sampling rate which are mentioned later in the discussion of choice of analog to digital converter.

3. Choice of Analog to Digital Converter

The choice of analog to digital converter to digitize the input signal is dependent on the size (voltage range) and the frequency of the input signal. There are three parameters to be considered in the choice of converters.

3.1 Voltage Range

One must insure that the voltage range of the converter is large enough to accept the input signal. If the range is not large enough saturation of the converter will occur causing incorrect recovery of the input signal. Figure 4A is an example of analog to digital converter saturation. A three volt peak to peak sine wave was presented to a converter with a range of plus one volt to minus one volt. The flat portion of the diagram indicates where the converter reached saturation (signal reached the one volt extremes). The



A) AN ANALOG VOLTAGE SIGNAL PASSED THROUGH A LOW PASS FILTER

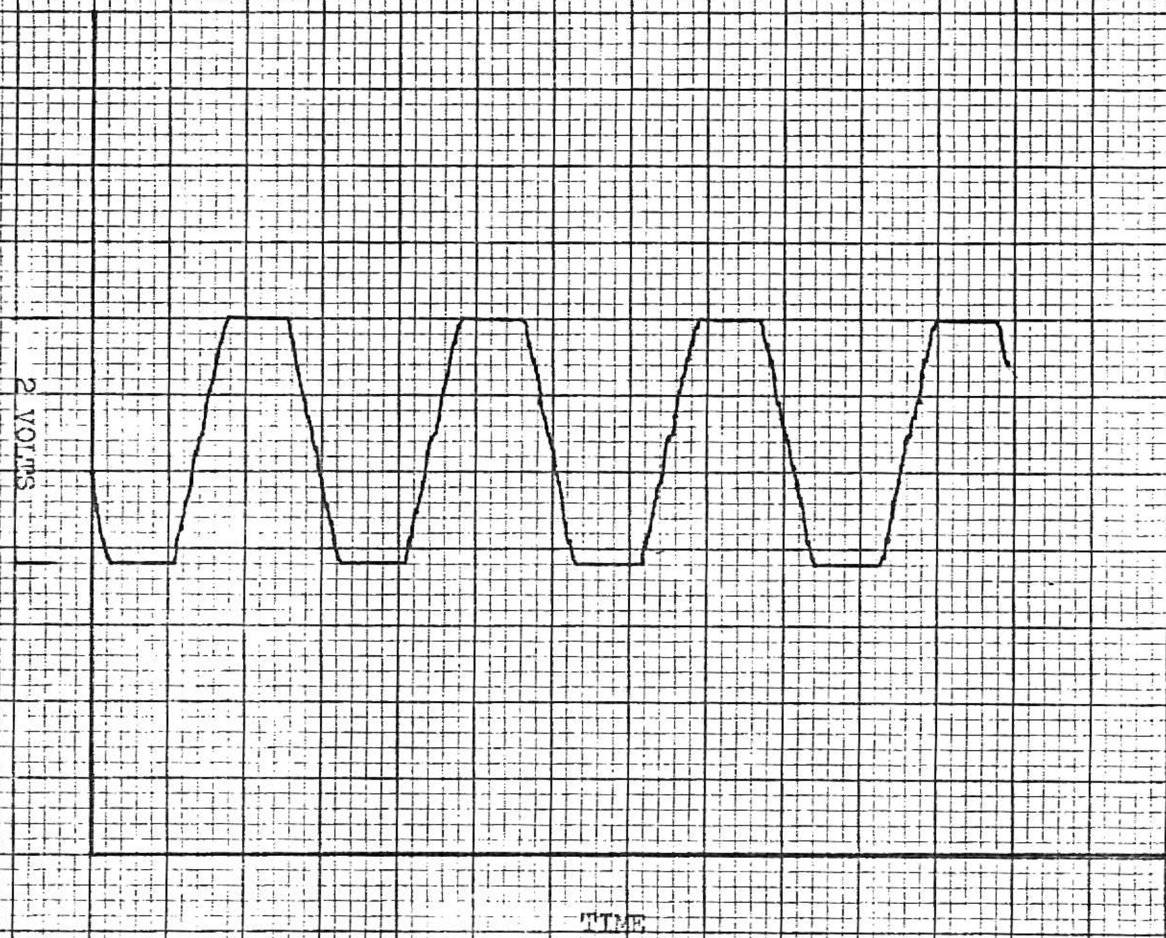
B) THE SAME SIGNAL--NOT FILTERED

FIGURE 3

dc level of the signal must be considered as well as the peak to peak ac amplitude of the signal presented to the converter. For example, a converter with a range of plus or minus one volt will accurately recover an input signal of two volts peak to peak if the dc level is zero. A dc level of other than zero will cause saturation on one side of the converter depending on the dc level. It is important then to regulate the dc level of the input signal to insure that a dc shift does not occur which will cause converter saturation. Figure 4B is a one volt peak to peak signal with a positive dc shift causing converter saturation on the positive side.

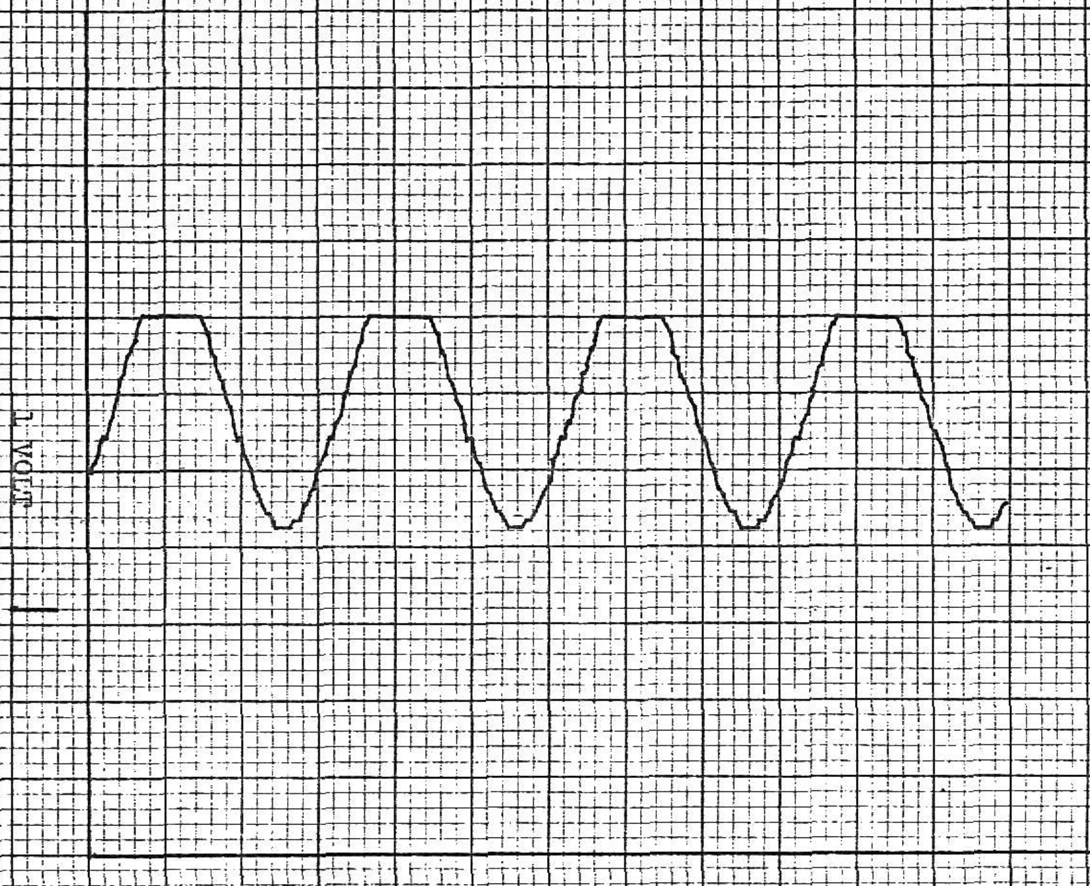
3.2 Accuracy

Another consideration, which is based on the frequency of the input signal, is the accuracy to which a given analog to digital converter can digitize an item of information. The input signal can take on any of an infinite number of values within the physical range of the signal. The converter, on the other hand, is limited to expressing a value in terms of one of a finite number of discrete values. The problem arises then when two successive samples of a continuously changing signal are digitized to the same value because the converter cannot distinguish between the values of the two samples. This would cause the digitized signal to "step" rather than to accurately represent a continuously changing signal. A slowly changing signal (low frequency) sampled at a high rate or a converter with too few bits of accuracy will cause this stepping to occur. Figure 5 is an example of a slowly changing signal sampled at a high rate causing "stepping." This problem may be solved several ways. If the range of the frequencies of interest is great, necessitating a high sampling rate which causes the low frequencies to step, one must acquire a converter with a greater number of digital bits. If one is sampling at a much higher rate than is required by the sampling theorem and stepping occurs, the sampling rate can be lowered until the



EXAMPLE OF ANALOG TO DIGITAL CONVERTER SATURATION--SIGNAL TOO LARGE FOR CONVERTER

FIGURE 4A

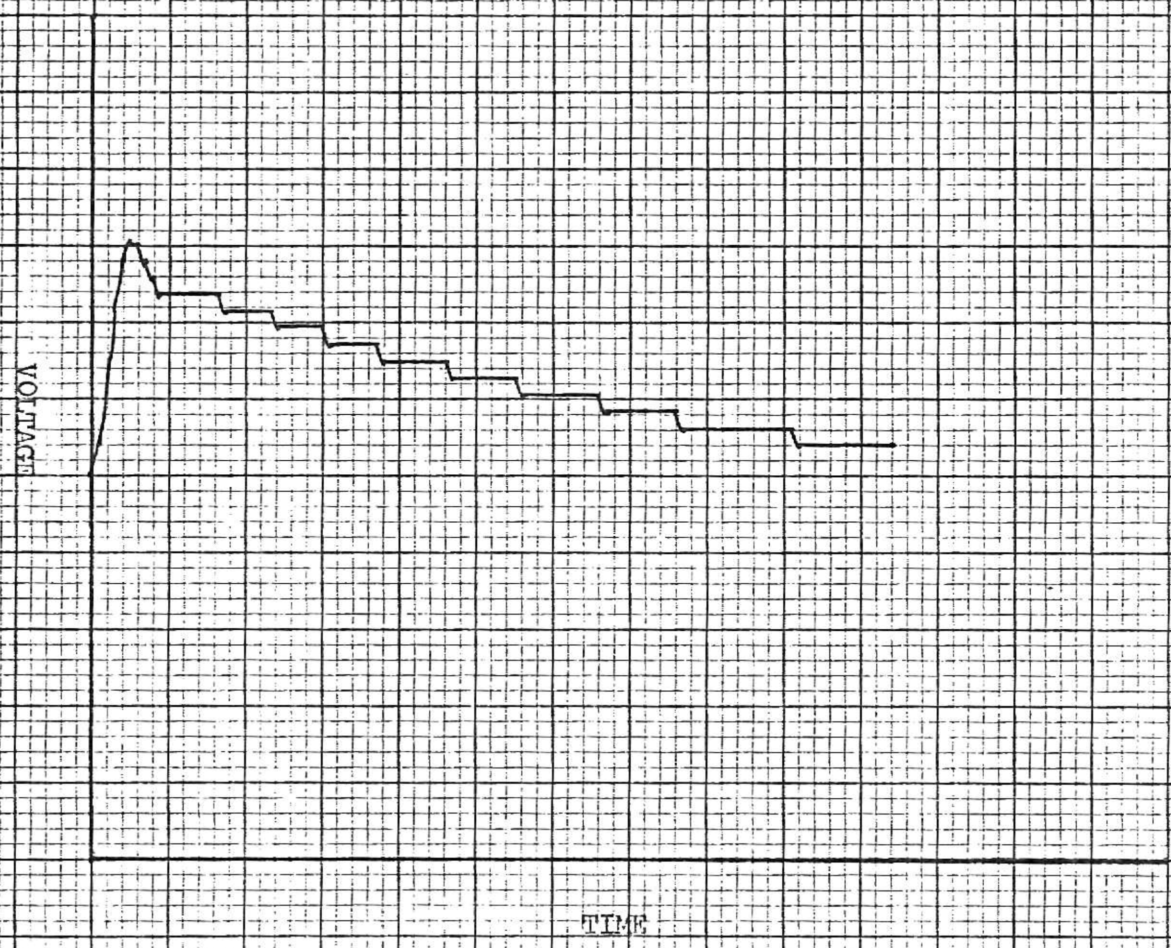


TIME

EXAMPLE OF ANALOG TO DIGITAL CONVERTER SATURATION--POSITIVE

dc SHIFT

FIGURE 4B



EXAMPLE OF SIGNAL STEPPING

FIGURE 5

stepping disappears without losing the frequencies of interest.

3.3 Conversion Time

The continuous input signal is changing with time thus the numerical measurement must also change. Converters require a finite interval of time to change from one value to another. One therefore cannot sample a signal at a rate greater than the time necessary for the converter to change values. This would become a problem only if an extremely fast sampling rate was required to recover the input signal since most devices today take only microseconds to change value.

4. Multiple Signal Sampling

In many digital measurement applications, more than one input signal is to be sampled. In addition to the considerations necessary in sampling one input signal, sampling multiple signals gives rise to some additional considerations.

Analog to digital converters are available with a multiplexer which allows sampling of multiple input signals. Converters which also have sample and hold circuits allow the simultaneous sampling of all input signals. When sampling is initiated all input signals are sampled simultaneously and the analog sample values are held in the hold circuits until conversion is initiated by the software. An additional problem arises when using a multiplexed converter which does not have multiple sample and hold circuits. The multiple input signals cannot be sampled simultaneously. There is a minimum delay between samples of successive signals of the conversion time of the converter. The maximum sampling rate of any one signal is then the product of the conversion time by the number of input signals. Due to the fast conversion time of most converters, this is not a major restriction on the sampling rate unless a large number of signals is to be sampled at a high rate.

5. Choice of Main Processor

There are many variables one must consider when purchasing a digital processor. No attempt will be made here to enumerate all variables. A few items which must be considered when choosing a processor to be used for digital measurement will be mentioned. These include processor word size and amount of core storage necessary. Both these items are dependent on the digital measurement application.

5.1 Processor Word Size

The processor word size must be compatible with the analog to digital converter used. That is, the processor word size must be at least as large as the converter word size, otherwise digital information will be lost in the transfer of information from the converter to the processor. If numerical operations are to be performed on the data, such as summing, the processor word size must be taken into consideration. The choice must be made to either acquire a processor with a large enough word size to allow for wanted numerical operations or realize that double word operations must be performed which increase operation time and core requirements.

5.2 Amount of Core Storage Necessary

The amount of core storage necessary for a particular application depends on the sampling rate used and the duration of signal sampling. As data is received from the analog to digital converter during sampling it must be temporarily stored in core until it is moved to a peripheral storage device. The amount of data stored in this manner and the core requirements of the software are the deciding factors in the amount of core necessary. This point will be discussed further in Part II, Section 2, Data Buffering.

6. Choice of Peripheral Storage Device

The choice of peripheral storage devices is dependent on the digital measurement application. The main considerations are access speed of the device and the amount of storage available.

6.1 Access Speed

The access speed of a peripheral storage device can become critical in digital measurement applications. In same applications the signal is sampled at a desired rate for a specified duration with a delay between sample onsets. In this type of application there may be enough delay to allow storage on a relatively slow device such as tape. If a signal is to be sampled continuously with no delay, and stored, a sampling rate must be chosen which allows enough time between samples to store the collected data. A device must be chosen then to fit the requirements of a particular application.

6.2 Amount of Peripheral Storage Necessary

The amount of data to be captured is the deciding factor in choosing the amount of peripheral storage necessary. If large amounts of data are collected the decision must be made to acquire high speed, large capacity, more expensive disc storage or slower, smaller capacity, less expensive tape storage. Storing large amounts of data on tape may require that the data tapes be mounted and dismounted during data collection. The amount of peripheral storage necessary depends on each particular application.

7. Choice of Output Devices

The minimum requirements for output devices depend on the digital measurement application. Generally, the minimum requirements are a printer and a plotter. The operator console may be used for printed output but greater output speed may be needed, necessitating the purchase of a high speed printer. A plotter, such as a Calcomp plotter, would be ideal for signal plotting although a much cheaper servo recorder could be adapted to the system giving limited plotting capabilities. Dot matrix printer-plotters are also available which give the capabilities of both high speed printing and plotting.

PART II

SOFTWARE CONSIDERATIONS FOR DESIGN OF AN ANALOG VOLTAGE DIGITAL ANALYSIS SYSTEM

1. Signal Preservation

In voltage signal measurement and analysis the peak to peak amplitude of the signal and the inflection point latencies are important. The amplitude and the time scale of the original signal must be preserved in the digital system.

1.1 Amplitude Preservation

An analog to digital converter receives a voltage input and digitizes that input to represent the original signal. The digital output from the converter is not in voltage units but is the voltage mapped onto the digital range of the converter. In processing the voltage signal to retreive the amplitude, the ratio of voltage input to digital output of the converter must be known so that the digital amplitude can be converted to the original voltage amplitude. In many applications such as measurement and analysis of polygraph data, the signal is passed through amplifiers prior to being presented to the converter. If this is the case, then to preserve the original voltage amplitude the conversion must also include the input-output ratio of the amplifiers. An example of amplitude conversion is included in Appendix I.

Since preservation of the signal amplitude is important, a method of conversion should be used which makes the conversion independent of the signal source. This can be accomplished by digitizing a signal of known amplitude from the signal source and having a software routine which figures the conversion ratio. If this is done, the software system is completely independent of the signal source; if the signal source is changed the software system is compatible with the new hardware.

1.2 Time Scale

As has been mentioned previously, reconstruction of the time scale is dependent on the sampling rate used. If the sampling rate is known, then to find the latency of a point one must only know the offset of that point from the zero point and the time between each sample. The accuracy to which the time scale can be recovered depends on the sampling rate used.

2. Data Buffering and Storing

The buffering of signal samples from the analog to digital converter into core prior to storage on a peripheral device is necessary in all applications of digital measurement. A buffer size must be used that is compatible with the measurement application and hardware available. If the signal is sampled in bursts of given duration and the hardware allows, the buffer should be large enough to contain the samples captured during each burst. This allows for storage to be accomplished during sample onset delays. If the signal is sampled continuously, it is best to have the sampling and buffer storage controlled under an interrupt driven system. This would allow for the storage of portions of the data between signal samples. The buffer size is dependent on the sampling rate used, the amount of core available for buffering, and the access speed of the peripheral storage device.

3. Operating Systems

The measurement of analog voltage signals is best suited to a dedicated computer system rather than a time-shared system. To preserve the time scale in digital measurement, the input signals must be sampled in real time. In some applications- the real-time analysis of the input signals is also necessary in order to immediately feed back information about the input signal. There are available computer systems which are well adapted to real-time signal measurement and analysis. Such systems allow implementation of real-time applications in assembly language as well as implementation of real-time applications in a high level language such as FORTRAN. Such systems generally have software available to perform real-time multiple signal measurement and analysis.

Most minicomputer operating systems do not perform the necessary task of synchronizing the various hardware operations such as the input and output of information. It is an extremely important programmer task to synchronize these computer operations.

4. Real-Time Acquisition Software

In real-time data acquisition there are some concepts which may be new to the high level programmer, such as a FORTRAN programmer. These concepts are likely to more familiar to an assembly language programmer. These include the use of software commands to directly manipulate the hardware. The control of the clock for timing and the converter for sampling are under direct control of the software. Synchronization of hardware operations is also under direct software control. Synchronization includes the concept of the wait state, an operation state in which the software is waiting for the hardware to perform its assigned function. Some wait states may be hardware controlled, such as the operation time of a converter. The hardware halts operation of any software until the conversion function is complete. Other wait states are software controlled through software wait loops. A function is initiated by the software, such as printing a character, and a flag on the device is checked and rechecked by the software until the operation is complete. When the hardware operation is complete then the software can continue operation. When a device is controlled under an interrupt system, the hardware function is initiated by the software and execution continues until the function is complete. When the device has completed its function, the hardware causes an interrupt which causes the execution of the software to be terminated and control passed to an interrupt handling routine which services the device before returning to the original software.

Before implementing a real-time data analysis system the programmer should become familiar with the concepts outlined in this section.

5. Interrupt Driven Systems

Sampling and storing of analog voltage input signals can be controlled under an interrupt driven system. In most applications the time between samples is relatively large. If data is sampled in bursts, the time between sample onsets is also available for operation of other software. The digitizing software can be operated as a foreground program to other software being executed. Sample onset and sampling can be initiated by the interrupt system. That is, the background program can be executing and be interrupted by sample onset and sampling and then continue execution between samples. The storage of the digitized data can also be controlled under the interrupt system, allowing for the storage of data between samples and during the sample onset delays.

In driving the sampling and storing of input signals using an interrupt system, the programmer has the task of setting priorities on the interrupts. For instance, in a digitizing system the sampling of data has top priority. When the data is being sampled no other task is more important. The setting of priorities is under software control. Some systems have an automatic priority interrupt option where the programmer may set or change the priorities and the hardware performs the task of following the priorities. In other systems the priorities must be handled completely by the programmer by turning the interrupts of devices on and off and servicing devices only when time allows. In order for it to be feasible to operate under an interrupt driven system, the hardware configuration must be compatible to such a system. There must be enough core and peripheral storage available to operate both sets of software.

6. File Systems

A filing system which allows for the direct access of any piece of information is extremely important in a digital measurement and analysis system. Files must be structured so that any piece of information or any set of data may be located with a minimum of effort. General filing practices of naming or numbering each set of data uniquely should be adhered to. The data itself should also be stored on the storage medium in a manner which facilitates easy access of each bit of information. Storing the data in block structured data files makes the locating of each bit of information trivial. For instance, if the data is sampled in bursts of known duration each burst may be stored sequencially in a data block. To locate any burst at a later time one need only know the sequence number of that burst. Enough said about filing systems. It must be remembered that after data has been stored, there must be some method available to later retreive that data.

7. Signal Analysis Software

Signal analysis software is part of the operating system software package of some systems. Software is also available through computer users societies and software support organizations of the manufacturer. This software is usually general purpose in nature, allowing the user to suit the software to his particular application. Following is a sample of the type of software generally available.

7.1 Signal Sampling

Signal sampling software packages allow the user to select and adjust the number of input signals, the sampling rate, and the storage medium to fit his particular application and system.

7.2 Signal Averaging

This software allows the sampling and averaging of multiple input signals under operator control.

7.3 Frequency Analysis

This is general purpose software for the displaying or plotting of the frequency spectra of multiple input signals.

7.4 Mathematical Operations

This is a software package of mathematical operations for manipulating the input signal. Included are high and low pass filtration, differentiation and integration, attenuation and amplification, inversion, addition of a constant, and bar graph plotting.

Glossary of Terms

Artifact	An artifact in EEG recording is any potential which does not originate in the brain.
Cut off frequency	In filtering, that frequency which appears at the three decibel level.
Decibel	Logarithmic ratio of signal attenuation, in voltage measurement 20 decibels is a ten to one ratio.
Digitizing	The process of representing each item of information as a unique group of digits.
Frequency Spectrum	The distribution of the frequencies in a signal.
Item of information	The value of the input signal at any given instant of time.
Latency	The time offset from the beginning of the signal to a specified occurrence (such as a maximum or a minimum).
Measurement	The determination of significant parameters of any real physical variable.
Octave	The frequency doubled from any given frequency (as from 30 hertz to 60 hertz).
Roll off rate	The rate of filter attenuation specified in decibels per octave.

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Appendix I

The parameters for an electroencephalograph digital measurement system.

The EEG digital measurement system described here is currently in use by Dr. Howard Shevrin, Senior Psychologist, The Menninger Foundation, Topeka, Kansas.

I. Input Data Specifications

The electroencephalogram as recorded from the surface of the head consists of rhythmical, slow, sinusoidal wave forms between 10 and 100 microvolts in amplitude (peak to peak) with a normal frequency range from .5 hertz to 30 hertz. Figure 6 is an example of an EEG signal.

II. Hardware Description

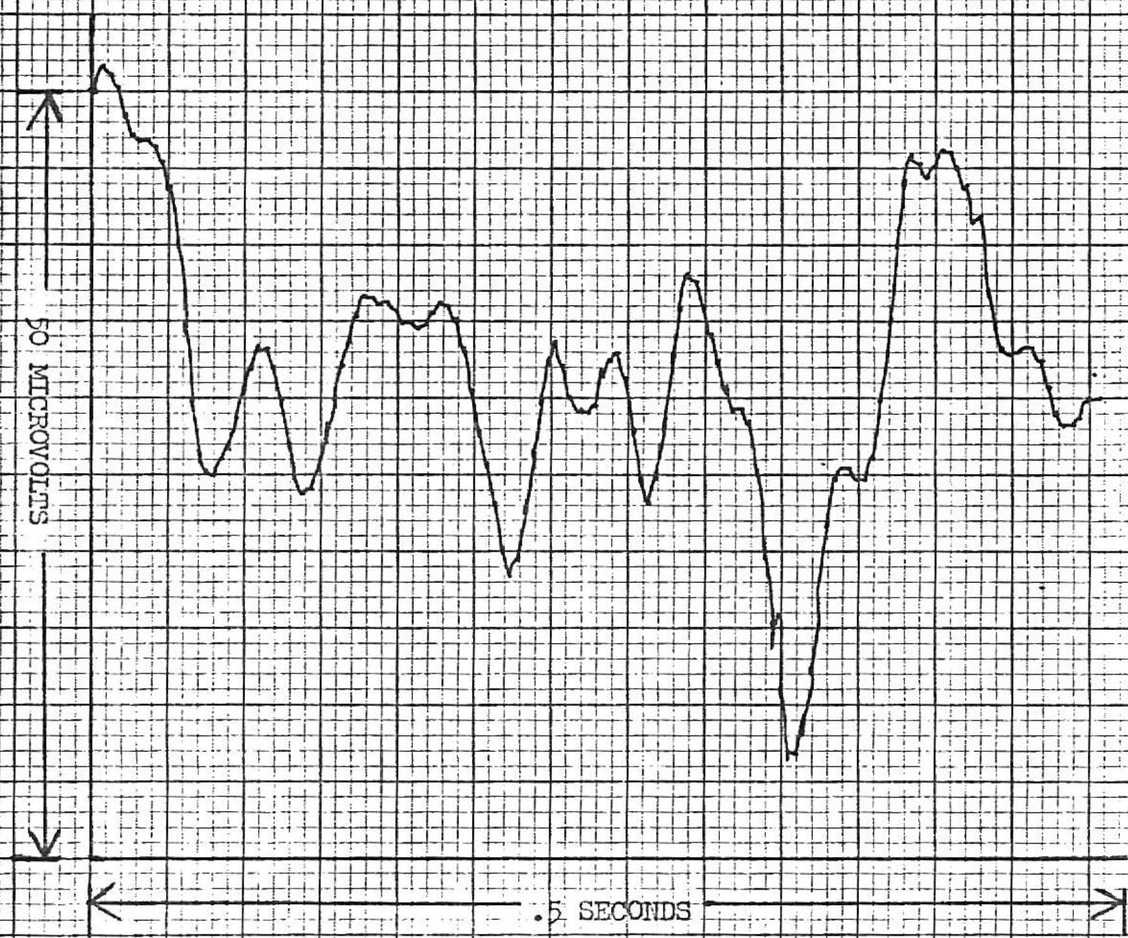
Figure 7 is the hardware configuration for measurement of the EEG data.

A. Electroencephalograph

The EEG currently in use is a Beckman Instrument, type CE, 8-channel electroencephalograph. The auxiliary output voltage is 2.8 volts per one inch pen deflection. The EEG calibration setting used is 7 millimeters pen deflection per 50 microvolt signal. This gives an output ratio of 16 millivolts per microvolt. The maximum signal then is 1.6 volts peak to peak.

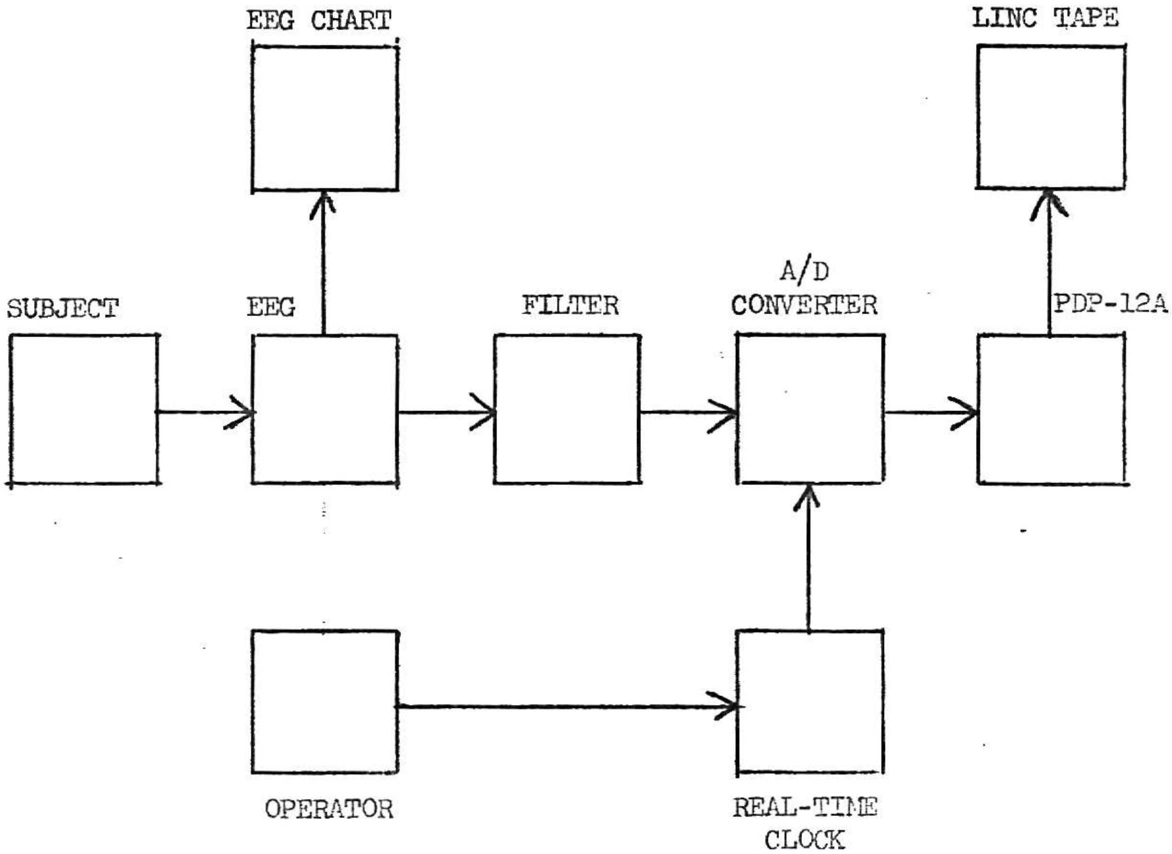
B. Data Filters

A high frequency noise signal is present in the EEG output, therefore, a filtering system must be used to eliminate this noise, yet preserve the EEG signal which has a maximum frequency of 30 hertz. The filter used is a low pass active filter with gain control buffered output. The cut off frequency of this filter is 30 hertz with a roll off rate of 24 decibels per octave. Figure 8 is a plot of



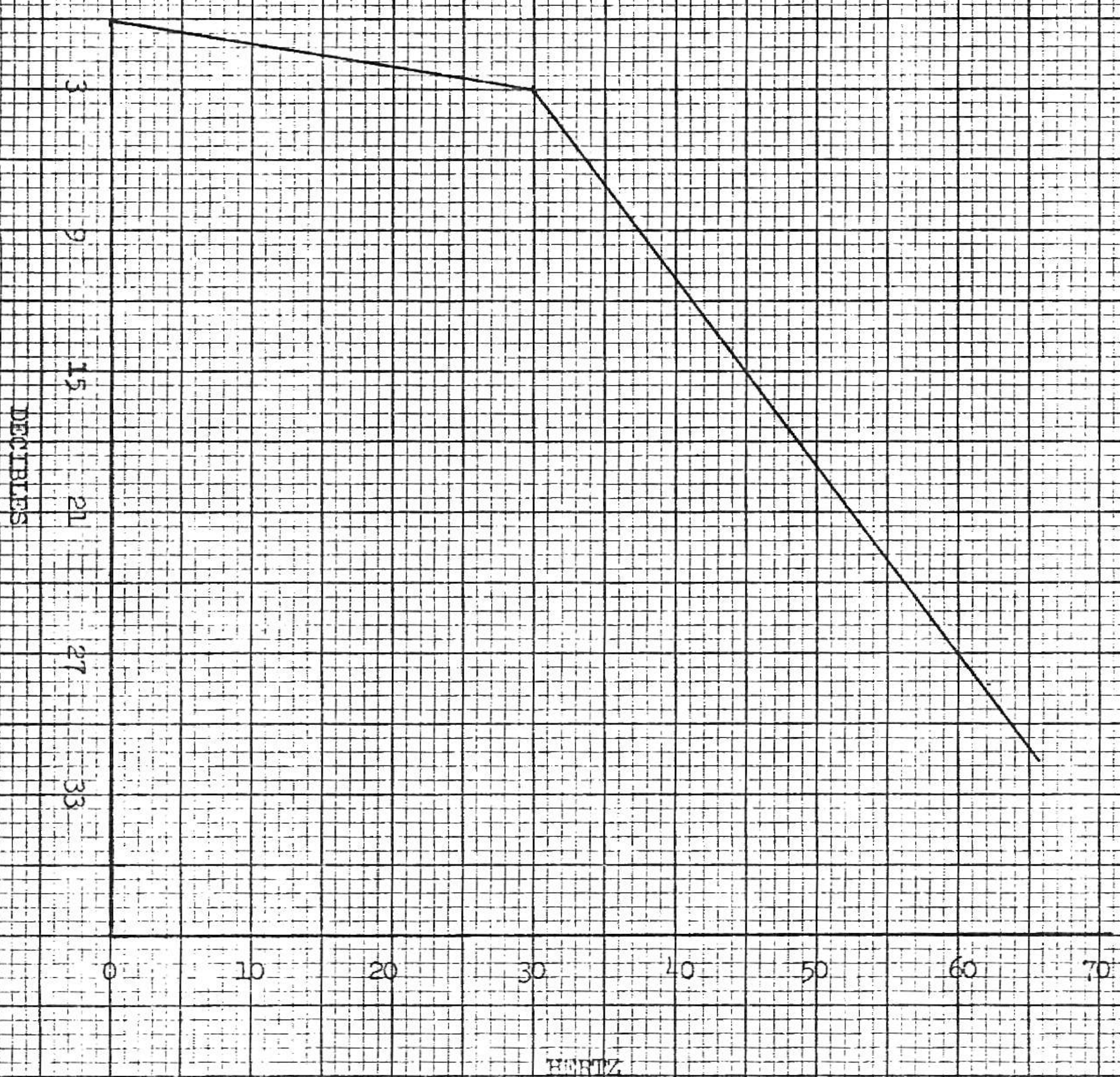
SAMPLE ELECTROENCEPHALOGRAPH SIGNAL

FIGURE 6



HARDWARE CONFIGURATION--EEG DIGITAL MEASUREMENT SYSTEM

FIGURE 7



CHARACTERISTICS OF LOW PASS FILTER:
CUT OFF FREQUENCY 60 HERTZ
ROLL OFF RATE 24 DECIBLES PER OCTAVE

FIGURE 8

the filter characteristics.

C. Analog to Digital Converter

An 8-channel Digital Equipment Corporation model AD-12 analog to digital converter with an input range of plus one volt to minus one volt (2 volt peak to peak) is used. The converter has a 10 bit digital range (plus 1023 to minus 1024) with a normal operation time of 18.2 microseconds.

D. Real-Time Clock

Sample timing is controlled by a Digital Equipment Corporation model KW12-A real-time clock. A clock rate of 100 hertz is used.

E. Digital Processor

The main processor is a Digital Equipment Corporation PDP-12A minicomputer with 8 K of core and a dual LINC tape drive for peripheral storage.

III. Software Description

Although a software package for digitizing and analyzing voltage input is available on the PDP-12, due to the method of data analysis in this application the decision was made to write a software package to fit this application rather than use the available software package. In this section an overview of this digital analysis application will be presented as well as the software features of the PDP-12 used in implementing this system.

A. Application

Six channels of EEG data are sampled for a duration of two seconds at a rate of 250 samples per second. Sampling onset is manually triggered by the EEG operator at a minimum of 5-second intervals. Each set of 1500 samples (hereafter referred to as a response) is stored on tape during the onset delays. Each trigger also presents a stimulus to the subject in the form of a picture flashed on a tachistoscope. There two different stimuli presented at two different speeds, thus there are four different categories of data collected. A full set of data consists of 36 each of the four categories or 144 responses. Data is stored consecutively by stimulus presentation on four reels of LINC tape. After a set of data has been digitized and stored on tape, information pertaining to the order and number of stimulus presentations and responses containing artifacts are entered by the operator. Using this information a data directory is constructed to be used by the analysis software in identifying each response on tape. This file contains the number of responses in each data category, the number of good (artifact free) responses by channel in each category and the data reel number and the tape location of the beginning of each response. In the analysis programs this directory is used to locate the correct data reel and beginning location of each response. Each of the six channels are located by an appropriate offset from the beginning of each response and the artifact flag word is checked and each channel either included or excluded from the analysis based on the state of this flag. The responses contained on each reel of tape are analyzed and a request is made through the operator console to mount the next consecutive data tape. Then analysis results are stored on scratch tape by category and printed when the analysis is complete.

There are two methods of analysis performed on each channel of each response. Both consist of finding a positive (downward) going wave and recording the latency of the beginning and ending of this wave along with its amplitude. In both methods data points 10 through 65 (40 through 260 milliseconds) are analyzed. The first method, which is called the peak method, locates the largest positive wave between these data points. The second method, which is called the sequence method, locates the first positive going wave which ends beyond data point 23 (92 milliseconds).

B. Implementation

Detection of the sample onset trigger pulse and sample timing are controlled by a Digital Equipment Corporation model KW 12-A real-time clock. This clock can be operated under program control at several clock rates and running modes depending on the use to be made of the clock. In this application the clock is used in two different ways. The clock is first used to sense the external trigger pulse to initiate sampling and then used to time the delay of 4 msec. between samples. The two-second duration is detected by keeping a tally of the number of samples made and halting sampling after 500 samples have been made. The clock counter is a 12-bit register which counts up to 7777 (octal) and then overflows on the next pulse. The overflow can be detected by an interrupt or a skip instruction. A buffer/preset register in the clock can be used to set the clock counter to a desired starting value and allow it to count up from that number to overflow. For timing then, the clock rate as well as the number of counts until overflow occurs can be software set. The mode of clock operation used for both detection of the external trigger and timing of sampling is the same but different clock rates are used. The mode used is a free running mode with preset. The buffer/preset register is set to the negative (2's complement) of the number of clock pulses until overflow. When the clock operation is initiated the clock runs at a selected rate, when overflow occurs the program detectable overflow flag is set or an interrupt pulse is generated and the contents of the buffer/preset register are transferred to the clock counter and counting continues. For detection of the external trigger pulse the clock rate is set to count from the external clock input and the b uffer/preset register is set to minus one (7777 octal) so that one pulse from the clock input causes overflow. The overflow flag is monitored with a skip instruction until overflow occurs indicating that a trigger pulse has been received. Occurrence of overflow initiates operation of the section of the program which samples the six input channels of the analog to digital converter. On receipt of the trigger pulse the clock rate and the value of the buffer/preset register are changed for timing the 4 msec. sample intervals. The clock rate is set to 10 kilocycles and the buffer/preset register is set to minus 400 (7160 octal). This causes overflow to occur every 400 clock pulses or every 4 msec. The overflow flag is monitored with a skip instruction and when overflow occurs the six data channels are sampled and stored in core. After 500 samples of each channel have been made, the data is stored on tape and the clock reset to detect the external trigger.

Sampling is accomplished using a Digital Equipment Corporation model AD12 analog to digital converter which consists of eight phone-jack external input channels, a multiplexer, and a ten-bit A-D converter. A single instruction selects the input channel, initiates the conversion, and transfers the contents of the converter buffer into the accumulater. Since there is no sample and hold function on this converter, sampling of the six input channels must be done consecutively. When sampling is initiated by the clock counter overflow, input channel 1 is sampled and then stored from the accumulator into core, input channel 2 is sampled and stored, etc. The 500 sample points of each of the six channels are stored in consecutive core locations. That is, the 500 sample points of channel 1 are stored followed by the 500 sample points of channel 2, etc. There are 12 locations prior to each sample which are zeroed for future software use. Each of the six channels then is stored in 512 consecutive core locations. This makes the buffering of the signal samples compatible with the LINC tape which is used for storage. During the sample onset delays the sample points from the six input channels are stored on tape.

The LINC tape system on the PDP-12 utilizes tapes which are preformatted into blocks of 256 12-bit words each. A tape reel consists of 512 of these 256 word blocks. Read and write tape operations are initiated by the software and all transfers affect the full 256 words of a tape block. For instance, in a write instruction the write tape demand is given along with a tape block number and a starting core location. The 256 consecutive words starting at the given core location are transferred onto the given tape block. Thus, all data is handled between tape and core in blocks of 256 words. In this application the data is buffered into blocks of 500 words for each data channel. Data is stored on two consecutive tape blocks for each channel or a total of 12 tape blocks for each response.

Once the data has been digitized and stored on tape and the information necessary to build the data directory has been entered by the operator, data analysis can begin. A 50 microvolt calibration signal from the EEG is digitized and stored on the last data tape of each set of data collected. This signal of known amplitude is used to insure that the amplitude conversion from digital counts to microvolts is consistent from data set to data set. Using the digital to analog converter in the PDP-12 display scope and the display commands, the calibration signal is plotted on a servo recorder. This signal is plotted with a known Y-range setting on the recorder. The amplitude of this signal in millimeters is entered to the analysis program which figures the correct conversion ratio for each set of data. In data analysis each response is read into core, the peak and sequence waves are found, and the beginning and ending latencies and amplitude of the waves are stored on scratch tape by category. When all data has been analyzed the analysis results are printed. A positive going wave is found by comparing two consecutive sample points and detecting changes in slope. When the slope changes from positive to negative then a maximum has been found. When the slope changes from negative to positive then a minimum has been found. When a wave has been found it is checked to determine if it fits the criteria for the scoring method and if so is saved as a result. If the wave does not fit the criteria then the search for another wave continues.

After the analysis of each response is complete, the data for each category and each channel is summed and the analysis is repeated on the resulting 24 summed waves. Since the PDP-12 has a 12-bit word size and the A-D converter has a 10-bit capacity, the summing of the data must be accomplished in double precision. Although this hardware configuration of the PDP-12 does not have double precision arithmetic instructions, instructions are available to perform double precision summing. The low or half of the sum is completed and if overflow occurs the overflow bit is trapped and can be added to the high order word. After the data is summed it is scaled by shifting right until the sum can be contained in one 12-bit word. This is done so that the software used to analyze individual response data can also be used to analyze the summed data. The conversion ratio is scaled so as to be compatible with the scaled version of the summed data.

A TUTORIAL ON DIGITAL MEASUREMENT AND ANALYSIS OF ANALOG VOLTAGE SIGNALS

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Abstract

This paper is a tutorial on accurate measurement and analysis of analog voltage signals for persons interested in the general field of digital measurement. Included is a discussion of the considerations necessary for a hardware system as well as the considerations necessary for a software system to accurately measure and analyze an input voltage signal. Possible applications of digital measurement and analysis methods include the measurement of biomedical variables such as electrocardiograph data and electroencephalograph data, business applications such as digital control of equipment or machinery or any other application where a voltage signal is to be measured.