no more than a few minutes. However, where data are being taken for very much longer periods of time, as would be the case in aircraft flight testing, several improvements are indicated, the most important being increased plotting speed.

There seem to be two paths that this development can follow. First, put the output on magnetic tape, instead of binary cards, in a form suitable for a tape-driven high-speed line printer. This would enable the printer to print all the characters for one line in one machine cycle, and also take advantage of the higher printing speed of the new printers. The second way would be to use a cathode-ray oscilloscope output for the *701*. This would reduce *701* output time, and leave as the only subsequent step the development and reproduction of the film. This method would have the further advantage of reducing the error presently introduced because of fixed time interval in the plotting.

In general, the system has been very satisfactory and has adequately fulfilled a definite need.

A PDM Converter

W. R. ARSENAULT

DIGITIZING Pulse Duration Modulated (PDM) data at a rapid rate and presenting it in a suitable form for data reduction has been a problem of data reduction centers for some time. The Magnavox Series *200* Converter is designed to accept PDM data recorded on magnetic tape, automatically digitize it, and record the digital information on the magnetic tape in a form suitable for input to a digital computer or other data reduction equipment.

The original PDM data are obtained by recording telemetered or ground data in the usual way during test runs. Using two intermediate tape units, the converter produces appropriate gaps in the digital output tape, making it compatible with such formats as that used by the International Business Machines Corporation (IBM) *701*. A data-tracking servo is incorporated in order to keep the digital output tape at constant density regardless of variations in the sample rate of the input tape. The servo also acts as a noise filter, producing a recording continuity on the final digital tape during periods of sporadic noise or long intervals of interrupted telemetered data.

Introduction

Pulse Duration Modulation has been used for some time now as an information carrier in telemetry systems. The method is to sample various analogue types of signals, usually appearing as a direct voltage, and converting this into a pulse, the duration of which is a function of the magnitude of the signal. This same result may also be realized by having a pulse, say positive, indicate the start of the duration and a second pulse, negative, indicate the end of the duration.

The system for which the Magnavox Series *200*, model *201* converter was designed, transmitted these pulses via a frequency-modulation system and eventually recorded the data on tape. Fig. 1 shows how these data appear on tape. The original sampling system contains a commutator that sequentially switches various direct voltages into a unit that generates the PDM data. This unit is called a keyer. The output from the keyer is a sequence of pulses, spaced equally in time but of varying duration. This shows up in Fig. 1 as pulses with equal intervals. In the particular system described the commutator has 30 segments. Of these 30, 28 contain data and two are left blank.

There are various ways of reducing these data to a usable form. Analogue methods have been used to scale and calibrate the samples and plot them directly. Digitizing the data allows processing by a digital computer. The general subject of processing these PDM data in a computer is covered by Lowe and Middlekauff.[1]

Conversion Problem Defined

The major design problems encountered were those in making the output tape compatible with the tape units and format of the computer with which it is being used. Specifications on both the input and output tapes will be discussed before discussing these problems further.

As pointed out in the "Introduction," the information recorded on the input tape originated in an air-borne keying system. In this sampling equipment, a commutator with 30 segments revolves at 30 revolutions per second providing 900 samples per second. The commutator scans sequentially the various instrument channels and provides the inputs to the keyer. The output eventually appears on a frequency-modulated carrier, telemetered to a ground station. On the ground it is converted to a pulse form and recorded on magnetic tape.

The ground recorder is an Ampex Model *309*. This recorder uses a 1/4-inch tape and two recording channels are provided, one for PDM data and one for frequency-modulated (FM) recording. In this instance, the major data are on the PDM channel with the FM channel used for timing markers.

During recording on the ground the tape runs at 60 inches per second. A pictorial view of the information as it is recorded at the ground station is shown in Fig. 1. The pulse interval from leading edge to leading edge is shown to be constant (within the specified tolerances) and corresponds to the interval from one segment to the next on the commutator. The pulse duration is a function of the parameter being measured at that time. There are 28 pulses of varying width on this tape corresponding to the sampling of 28 different parameters by the commutator. Two segments are left blank in order to define a single frame or commutator rotation. Timing in Fig. 1 is for playback at 15 inches per second.

This magnetic tape now becomes the input tape to the converter. Specifications as they apply to a single pulse are shown in Fig. 2. The nominal pulse interval of 0.067 inch is derived from 900 samples per second being recorded at 60 inches per second. This interval may

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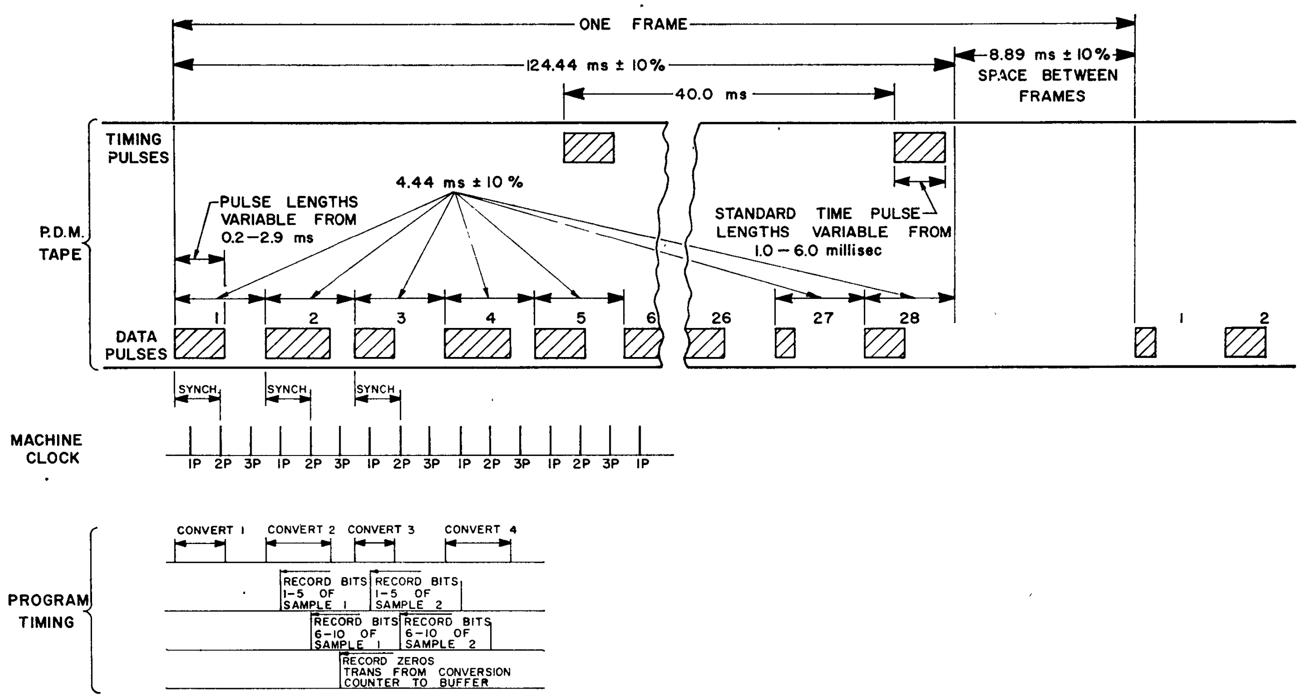


Fig. 1. PDM input tape and machine timing

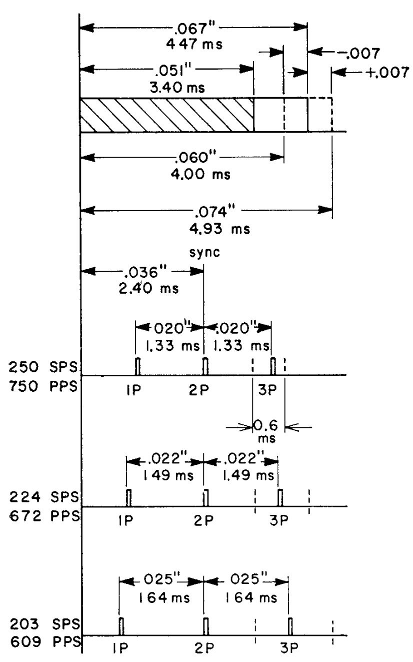


Fig. 2. Specifications of input sample pulse. Times are for data playback at 15 inches per second

vary ±10 per cent over a long period of time due to variations in the commutator speed; thus the tolerances of ±0.007 inch. The longest allowable pulse width, representing the sampled information, is 0.051 inch.

Fig. 2 also gives the pulse intervals in time when the input tape is played back at 15 inches per second. In the lower portion of the drawing the relation of the 3-period converter clock is shown and will be explained later.

In the particular application being discussed here, the output tape from the converter must be compatible with an IBM 726 tape handler. This is a 7-channel 1/2-inch tape with a format shown in Fig. 3. Six rows of six tracks constitute the 36-bit IBM *701* word. The seventh channel is a parity check channel; the information stored here is such that the sum of the seven digits in one row is always odd.

The 36-bit IBM word may be divided into two half words. The format of Fig. 3 shows a digitized sample recorded in each half word. The next word is adjacent to this one as is the following word. There is no discontinuity between these words, each being identified as a group of six rows. At predetermined intervals there is a 1-inch intra-record gap. The interval between intra-record gaps defines the amount of information to be read into the high-speed memory of the data processor during one reference to the IBM *726* tape handler. The density of the output tape must be 100 bits per inch with an allowable variation of only ±3 bits per inch.

With the specifications for the input and output tapes cited, the main problems for the converter can be defined. These may be broken down into two categories:

1. Operating the converter in synchronism with the input tape, the output digital information must be recorded at a constant density, although the input sample rate may vary as much as ±10 per cent.

2. A 1-inch gap must be inserted in the digital output tape, at predetermined intervals, while accepting continuous input samples.

Elaborating on these two points, it is necessary to keep a machine clock in synchronism with the incoming data samples so that the conversion and recording in digital form can keep in step and not lag behind these incoming data. With the machine clock, and, therefore, the recording rate, varying, it is necessary to vary the speed of the output tapes so as to record a constant information density.

The second item mentioned was that of inserting a 1-inch intra-record gap in the final tape without loss of the input data. The two blank channels, 29 and 30, do not allow enough dead space to stop and start the input tape, even with the fastest digital tape handler. Further, these digital tape handlers do not have the flutter and wow characteristics required to play back the analogue input tape.

Other functional requirements of the converter were relatively easy to solve. In previous discussions, it has been pointed out that several of the analogue samples are carried as calibration and scaling factors. All information within a frame, 28 samples plus two blanks, is referenced to these scaling factors. Requirements set on the converter do not include any absolute reproductions but only relative accuracies. The speed of the input tape must be kept constant over several frames so that there is no speed change within a frame. If these specifications are met for playback, then the samples can be treated as time duration pulses from the playback tape rather than the physical distances that they are.

System Description

The first problem outlined in the previous section, that of keeping the conversion in step with the incoming data and keeping the digital recording on the output tape of constant density, was solved by incorporating a data tracking servo. This was designed to keep the converter clock synchronized with the incoming data, with a minimum of phase error. As the incoming sample rate increases, the machine clock increases accordingly. The machine clock directly controls both the conversion to digital form and the recording, thus keeping these functions in step.

A simplified block diagram of the converter is shown in Fig. 4. The relation between the incoming data, the machine clock, and the machine program is shown in Fig. 1. There are three machine clock pulses for each data sample. This corresponds to recording the three rows of information per sample on the digital tape.

The PDM data are routed to both the data tracking servo and the conversion counter. The data tracking servo uses the leading edge of the sample as synchronizing information. The leading edge of the data pulse is delayed a fixed amount and called the "sync" pulse. The machine clock $2P$ is kept in line with the sync pulse by the servo.

Fig. 5 is a simplified diagram of the data tracking servo showing the phase error detection supplying an input to the operational amplifier. The delayed leading edge pulse is compared with $2P$ of the machine clock. If there is a leading phase error, machine clock ahead of data, a positive pulse of width equal to the magnitude of phase error is input to a smoothing network. The output of this network, and input to the operational amplifier, is a function of the amount of phase error and is greater the larger the error. This signal is inverted in the amplifier and appears as a negative going signal. This makes the multivibrator bias more negative and reduces its frequency.

If the machine clock lags the input data, a negative pulse drives the amplifier, and inversion in the amplifier increases the multivibrator frequency. There are two paths for the error signals through the amplifier. The first, through the capacitor, provides a fast action phase correction and the second, through the resistor, gives an integrated frequency correction. Both are needed since it is possible to have a phase error without having a frequency error.

Referring again to Fig. 2, a more detailed drawing of the input pulse timing and the machine clock variation as a function of input pulse interval is given in the lower section. The tracking servo is designed so that it lines up the sync pulse, which is the leading edge of the sample delayed 2.40 milliseconds, and the mid-clock pulse $2P$. As the input sample interval changes from minimum (4.00 milliseconds), to nominal (4.47 milliseconds), to maximum (4.93 milliseconds), the position of these clock pulses is shown relative to the input sample. In all cases the $3P$ pulse is approximately centered in the interval between the end of the longest pulse width and the leading edge of the next sample. $3P$ is used to gate the conversion counter, and centering $3P$ in this manner allows correct conversion of data over the greatest range of phase error between the machine clock and the incoming data.

The output from the operation amplifier also controls a second multivibrator which in turn controls the capstan speed of the intermediate tape units. As the input data rate increases, and therefore the machine clock, the capstan speed increases accordingly. This guarantees that the digital information recorded on these units will be of constant density regardless of the variation in the input sample rate.

The data tracking servo also acts as a noise filter providing a continuity of digital recording during periods of sporadic noise or temporary periods of complete absence of data on the input tape. The machine will continue to record zeros during an absence of data at the input at a repetition rate indicated by the most recent input samples.

Referring to Fig. 3, the PDM data go into a conversion counter. The conversion counter is a straight 10-digit binary counter. The leading edge of the input sample opens a gate and allows pulses from a crystal controlled oscillator into the counter. The pulses are counted until the trailing edge of the sample closes the gate. At this point a digital number proportional to the width of the

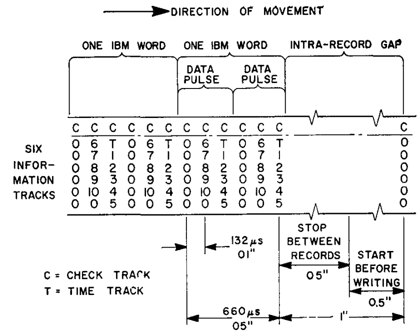


Fig. 3. Output tape format. This figure illustrates format for two channels of data. "1" is the most significant digit of a number; "10" is the least significant. The third row except for check track contains zeros in all cases

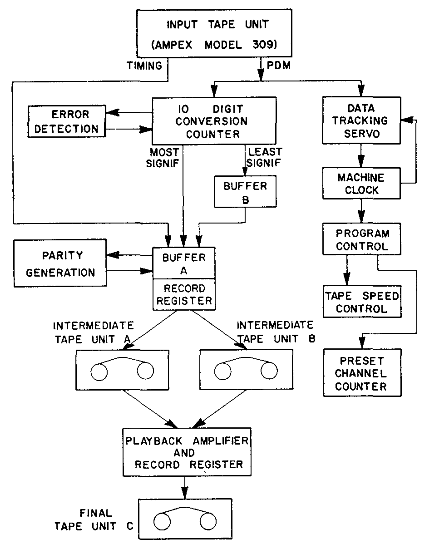


Fig. 4. Simplified block diagram

particular sample converted is in the conversion counter. This is transferred to a buffer register prior to the appearance of the leading edge of the next sample. During the time the second sample is converted, the first is recorded on the digital tape. This program sequence is depicted in Fig. 4. The format of the output tape (Fig. 2) shows the first five digits recorded in one row and the second five in the second row. This recording takes place at machine clock times $1P$ and $2P$ respectively. The third row is left blank, recording only zeros, making up the IBM 18-digit half word. The seventh track contains the parity digit. There is a parity bit for each row and each is generated in the buffer just prior to recording of that row.

The format of the output tape allows one digit for a timing bit (T). If a timing mark appears on the FM track of the input tape during the time a particular

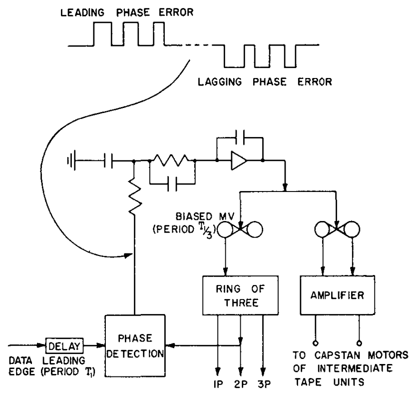


Fig. 5. Data tracking servo

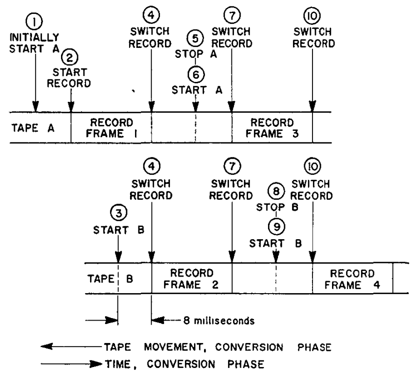
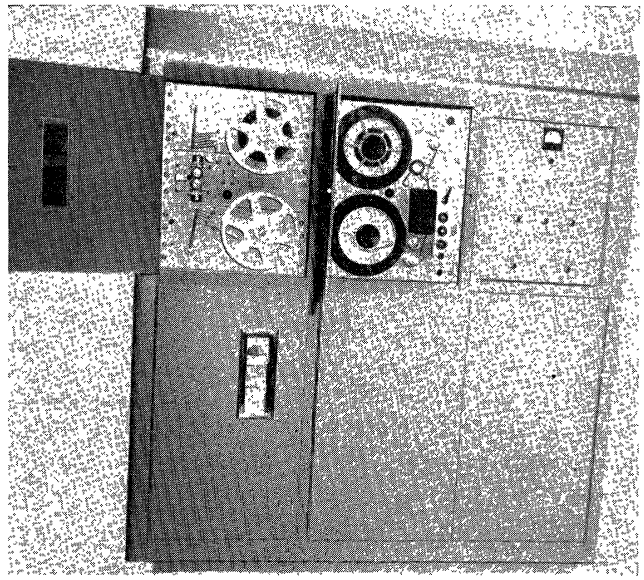


Fig. 6. Data arrangement on intermediate tapes. Note: Numbers in circles indicate sequence of events in time

Fig. 7. Front view, model 201 converter



sample is converted, a one is recorded in this spot along with the digital equivalent of the sample.

To this point in the discussion nothing has been said about inserting the gap and little has been said about the digital tape units. The system block diagram shows the information from the record register going to two intermediate tape units. The two units may be thought of as a large buffer storage. During the conversion process, called phase 1, the digital information is recorded in blocks on these units, alternately, until all of the analogue samples on the input tape have been converted. Then these tapes are played back, recording on a final tape unit *C*. This final unit has the information in a form required by the IBM *726*.

When the program is first started—the beginning of phase 1, the converted data are recorded on tape unit *A*. A preset channel counter is used to determine the number of samples to be recorded per record frame. The number of samples per record is selectable for a particular run and is at the discretion of the operator. It is dictated by the amount of memory space in the data processor available for 1-read reference to the tape units. A typical number might be 1,500 samples or 750 words. When information from the channel counter indicates that a gap is to be inserted in the final tape, the recording is switched from unit *A* to unit *B* without interruption, unit *B* having been started just before the record switch. Unit *A* is stopped shortly after the switch to *B*. When the next gap is to be inserted, the complementary action takes place, switching recording back to *A*. This process of switching back and forth continues until the input tape is exhausted of data.

The arrangement of the data on the two tape units after the first phase of conversion is completed is as shown in Fig. 5. All of the odd record groups appear on tape unit *A* while the even groups appear on unit *B*. Although only four frames are shown in the drawing, the quantity of data is limited only by the output tape capacity. The total amount of information converted is dependent upon the quantity of analogue information on the input tape.

When the conversion processes are completed, the intermediate tape units contain all the digital information. This is in the same format as that required by the output tape and depicted in Fig. 2. As discussed, the record groups, which consist of many converted samples, are sequentially arranged in record groups on alternate intermediate tape units. It is now necessary to collate the record groups and insert the 1-inch gap while recording on the final unit. When phase 2, playback mode, is started, the intermediate unit last recorded starts in reverse. A gap-sensing device indicates when the end of a record group is reached. This immediately stops the running unit and starts the opposite unit. This then plays back until a gap is sensed on this second unit. It will be remembered that, during the conversion phase, when recording was switched from one unit to the other, there was a delay before stopping the first unit. This delay created a blank gap between groups shown in Fig. 6 and is of the correct amount to produce a 1-inch gap in the final tape.

The process of alternately playing from each intermediate unit continues until both units run out of information. When all digital information is read, both intermediate units run back and are automatically stopped by a photocell sensing a clear leader.

The intermediate units are played back in the opposite direction from which they were recorded. The digital information is read and simultaneously recorded on the final unit *C*. This implies that the final unit must run in the opposite direction from which it normally will be used in order that the data be intelligible. For this reason, the final tape unit *C* is made to run out blank tape during the conversion phase. When it is time for phase 2, the correct amount of tape is on the take-up reel and the tape rewinds as it is recorded upon.

Machine Operation

All controls necessary to operate the converter are brought out to a panel on the front of the cabinet. Here the alternating and direct voltages may be turned on. Both a-c and d-c blown fuse indicators are here.

Once the alternating current and direct current are turned on and proper warmup time is allowed, the machine is ready for operation. The external Ampex unit must be loaded with the tape containing the sampled data; a clear leader of several feet should be threaded so that the tape unit is up to speed when the data are read. The final tape unit *C* is loaded with 1/2-inch tape.

Pushing the "start phase 1" button automatically starts the external input unit as well as all three tapes on the converter. When data arrive on the input tapes, intermediate tape unit *B* stops and converted data are recorded on unit *A*. Recording continues on *A* until the present-channel counter indicates that recording should be switched. Unit *B* starts and recording is switched to *B*, unit *A* stopping 180 milliseconds later. During this time unit *C* is running out tape, the amount that will be needed to hold all of the converted data on one tape.

The cycling process of recording on unit *A* or unit *B* continues until all of the PDM data are converted. At this time the operator pushes the "stop" button. All tape units stop, the intermediate units delayed only enough to record the last converted sample. Either tape unit may be recording when the stop button is pushed and it may be anywhere in a record group.

The operator now pushes the "start phase 2" button which starts the playback process and the recording on the final tape unit. Initially, tape unit *C* is the only one that starts. This runs for approximately 8 seconds producing a 1-foot end-of-file blank required by the IBM *726*. At the end of the 8-second period, the last intermediate unit recorded upon during phase 1 starts in the reverse direction. There is a 1-to-1 correspondence between the data on the intermediate units and that on the final unit. Digital information is read directly from the intermediate units and recorded on the final unit.

The playback continues from the unit started at the beginning of phase 2, say unit *B*, until a gap is sensed. Playback is immediately switched to unit *A* and unit *B* is stopped. Unit *A* now plays until the gap is sensed on this unit, and playback is switched again. This process continues until all of the record groups on the intermediate units are played back. The tape units are stopped automatically by photocell sensing. Tape unit *C* runs out an end-of-file gap and is then read for processing on the IBM *701*.

Conversion during phase 1 proceeds at the rate of 224 samples per second. The playback during phase 2 proceeds at twice this rate making an average conversion

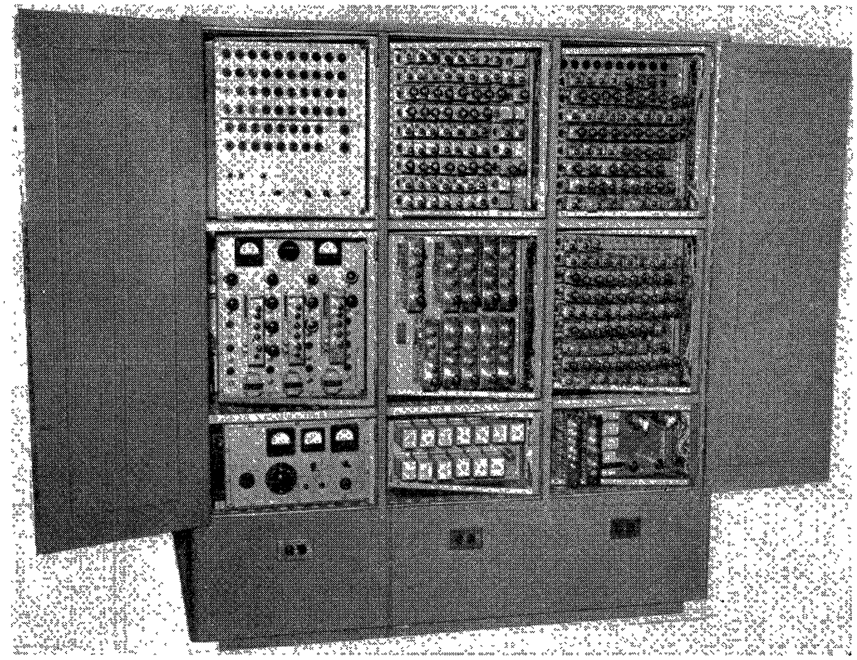


Fig. 8. Rear view, model 201 converter

rate of approximately 150 samples per second. The amount of conversion is limited by the capacity of the output tape and for a 2,400-foot reel this is over 900,000 samples.

Machine Design

A front and rear view of the converter is seen in Figs. 7 and 8 respectively. In the front view, the three tape units associated with the converter are shown. The two units on the left are the intermediate units and the center unit is the final unit *C* On the right upper is the control panel. The meter shows the output from the servo amplifier. Below it is an adjustment for removing any drift in the amplifier before operation is started. In the lower center and right are drawers containing the d-c power supplies.

The rear view shows the general approach of constructing all circuitry on plug-in units. This is designed as an aid in trouble-shooting and routine maintenance. The multitube units in the center are the playback amplifiers for the intermediate tape units. The panel in the lower left is for metering alternating and direct voltages and making adjustments in some direct voltages. These adjustments are used for marginal checking during routine maintenance. The panel in the upper left is a test panel where the majority of the plug-in units may be checked for operation independent of the machine proper.

There are 333 tubes in the machine of which a majority are *5670*'s used in trigger circuits and *7AK7*'s used in gated pulse amplifiers. The record tubes are *5687*-type. The converter also uses some 900 diodes of which the majority are *IN38A*'s.

The power supplies use germanium rectifiers and the total power consumption, a-c and d-c, is approximately 3.5 kva.

Conclusion

The foregoing is a discussion of a specific converter designed to do a specific task. The speed of conversion was decided to be 1/4 that of real time because of the short flights involved and because of the 10-digit accuracy desired. For this conversion rate the fastest trigger in the conversion counter operates at some 300 kc. Further, the output tape format was dictated by the data processor with which it was to be used.

The machine is versatile in that both the conversion rate and the output tape format may be modified to meet the needs of other data processing centers. Further it is possible to include editing features that will control the conversion process so that only sections of the PDM tape are converted. This feature would be particularly valuable where data from long flights are recorded but only certain sections are of interest.

Reference

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An Improved Multichannel Drift-Stabilization System

P. G. PANTAZELOS

AT PRESENT, all but the smallest d-c analogue computers employ some method of drift stabilization to reduce drift at the output of the computing amplifiers. The method most often used is called chopper stabilization.[1] With this method, some drift-free gain is added to the forward loop of a d-c feedback amplifier. If the added drift-free gain is placed in the loop ahead of the primary sources of drift, the steady-state drift with stabilization is equal to the drift without stabilization divided by the amount of drift-free gain added. The required drift-free gain can be achieved with a chopper and an associated stabilization amplifier.

By the technique of single-channel chopper stabilization, excellent drift stability can be obtained, particularly with well-built and well-shielded choppers such as the Leeds and Northrup unit. Unfortunately, during a 5-year period, experience with a large number of choppers at the Dynamic Analysis and Control Laboratory (DACL) at the Massachusetts Institute of Technology has shown that maintenance of these choppers is necessary after their first year of operation, and, as the choppers become even older, they must be maintained more and more frequently. Also, the amplifiers associated with each chopper must be checked periodically. In a large installation with more than 200 computing amplifiers, such maintenance presents a problem. Another difficulty is that these choppers are bulky, and they, together with the added amplifiers, prevent the construction of smaller computing amplifiers. These disadvantages of single-channel chopper stabilization are avoided by using multichannel drift stabilization.

Multichannel drift stabilization[2] is an extension of the chopper-stabilization technique. In the multichannel system as shown in Fig. 1, one stabilization system is time-shared by a group of d-c amplifiers, thus effecting a reduction in equipment size, cost, and maintenance. Unfortunately, the multichannel systems in use at the time that work began on the DACL system were inferior in some respects when compared with single-channel systems. For example, sufficient gain could not be achieved in the common stabilization amplifier to eliminate the need for a balancing adjustment in the computing amplifiers and in the stabilization amplifier. Crosstalk existed between channels, particularly if one channel was badly overloaded. Also, none of the systems at that time incorporated overload indicators that operated at the incidence of an overload and remained on after the overload until reset by the problem operator.

The design techniques described in this paper extend the usefulness of a multichannel system by eliminating some of these defects. The problem of eliminating crosstalk between computing positions without sacrifice of stabilization gain has been solved by the use of intermittent feedback and self-biased diodes, and the problem of providing a useful overload indicator has been solved by sampling the size of the signal in the stabilization system and by igniting gas tubes when an overload exists.

These techniques are now incorporated in a multichannel system in a small computer at the DACL. The computer has 30 d-c computing positions all of which are drift-stabilized with one commutator and one stabilization amplifier. The computing amplifiers require about 1/6 the volume of the older, single-channel chopper-stabilized amplifiers. The maintenance required is negligible when compared with the older units.

The response speed and the transient characteristics of this multichannel system as well as a statistical evaluation of the drift encountered in the DACL computing positions have been described elsewhere.[3]

Theory of Operation

In the multichannel system of Fig. 1, one stabilization amplifier is used to stabilize the 30 d-c amplifiers with the aid of a single commutator that samples the summing-point voltage of each amplifier in turn. Any direct voltage present at the summing point of an amplifier is applied to the stabilization amplifier as a pulse occurring at the repetition frequency of the commutator. These pulses, after being amplified, essentially without drift, and inverted in the stabilization amplifier, are channeled to the same d-c amplifier by the output section of the commutator and applied as a stabilization voltage through a smoothing filter. The computing amplifiers are conventional high-gain d-c amplifiers.

The stabilization circuit is used also for overload indication. When any d-c amplifier in the computer is overloaded, a large d-c error voltage appears at its summing point. This voltage is sampled by the input section of the stabilization switch, and the resulting pulses are amplified by the stabilization amplifier (see Fig. 1). Whenever the pulses in the stabilization amplifier exceed a predetermined level, they trigger a monostable multivibrator. The large, positive output pulses from the multivibrator override the normal output of the stabilization amplifier and are applied through the output section of the commutator to the overloaded d-c amplifier. In the d-c amplifier, where the large size of these pulses distinguishes them from a normal stabilization output, they are detected and used to trigger an overload indicator.

The Stabilization and Overload-Indication Unit

Fig. 2 is a schematic diagram of the stabilization and overload-indication unit circuitry. Tubes *V1*, *V2*, *V3*, and the first section of *V4* constitute the amplifier section. This is a conventional d-c amplifier. To eliminate drift in this amplifier, its gain is reduced between pulses to less than one by briefly closing a feedback path, as shown in Fig. 3. Each time the input grid of the stabilization amplifier is grounded, the charge on capacitor *C* in Fig. 3 is adjusted to the value required to keep the quiescent amplifier output very nearly at ground potential. No energy-storage elements appear in the stabilization-amplifier circuit during the time it receives signals from the computing-amplifier error points. The rotor of the input section of the commutator is wider than the rotor of the output section in order to ensure that the input be grounded whenever the stabilization-amplifier feedback loop is closed; therefore, crosstalk between adjacent

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