NATIONAL RADIO ASTRONOMY OBSERVATORY
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To: Scientific Staff

From: Marc Vinokur

Subject: Output System for a Filter Multichannel Receiver

INTRODUCTION

The use of a conventional multichannel hydrogen line receiver with the 300-foot telescope entails the necessity of recording into digital form a great number of data simultaneously and to do it quite often. Two types of problems are involved: First of all, the digitalization system itself, which must convert the output signals into proportional numbers stored on a convenient support (paper tape, magnetic tape, or cards, for instance), has to be fast enough and to meet the requirements of simplicity, reliability, and accuracy at a reasonable price.

Then, since the quantity of data to be handled is extensive, it is desirable to reduce it as much as possible, firstly by using the slowest rate of sampling compatible with the complete, or almost complete, utilization of the available information and, secondly, by limiting the number of significant digits for each measurement at an appropriate minimum.

These points were particularly important when construction of a 100-channel receiver based on the classical filter principle was being considered. The system described below was designed at that time to meet the appropriate specifications. It is consequently suited for the digitalization of a 20-channel extragalactic receiver, which is now being planned. Since, by modifying a few elements in this case, the economics should be negligible, it has not seemed worthwhile to reduce the performance of the system.

After a general description of the proposed output system, we will study in detail its requirements, and as will be seen, a careful choice of both the rate of sampling and the precision of measurement will not only allow us a reduction of the number of data to its minimum, but also will simplify to a certain extent the output system itself.

We will conclude with a few technical data and an estimate of the price of the complete system proposed.

I. PRINCIPLE OF THE PROPOSED SOLUTION

Due to the type of antenna used, the observations will consist of transit scans. As far as the signal alone is concerned, the sampling theorem, when applied to the antenna equivalent low pass filter whose spatial frequency cut-off is $\frac{d}{\lambda}$, states that the transit curves can be entirely restored by only the knowledge of discrete values equidistant by no more than $\frac{1}{2\frac{d}{\lambda}}$ (d is the diameter of the antenna, λ the observed wavelength), (1) (2). A similar result can be applied to the usual case where signal and noise are mixed. A first approach to establish it might be to use the same sampling theorem for the noise: If its spectrum, after filtering by a low pass filter has a bandwidth strictly limited to a value B, then a given record of noise is completely known with the data at discrete values distant of $\frac{1}{2B}$. Therefore, when signal and noise are superimposed, all the information they contain is specified by sampling their values at the one of the two rates which is the highest. Actually, the preceding is only a rough estimate since, firstly, the bandwidth of the noise is generally not strictly limited and, secondly, what interests us is not having the ability to restore signal function and noise function, but only the former by reducing to its minimum the error due to the latter.

These points, which will be considered later on, have been discussed elsewhere, (3), and an exact determination of the useful rate of sampling proposed. Anyhow, the important fact remains that both from the points of view of the signal and the noise, the recording of discrete values allows us to use almost all, if not all, of the available information.

An immediate consequence concerning the output system is that it is then possible to use only one channel of digitalization and recording for the whole receiver. The advantages of such a solution are not only in terms of simplicity and economy, but also in terms of overall accuracy since the same "output meter" is used for all the channels.

The proposed system comprises essentially (fig. 1) an analog-to-digital converter which converts the output voltage of a channel to a proportional number of pulses, a counter which displays this number in the desired code, and a tape punch. A stepping switch sequentially scans the channel outputs. The block diagram of the complete system is shown in figure 2.

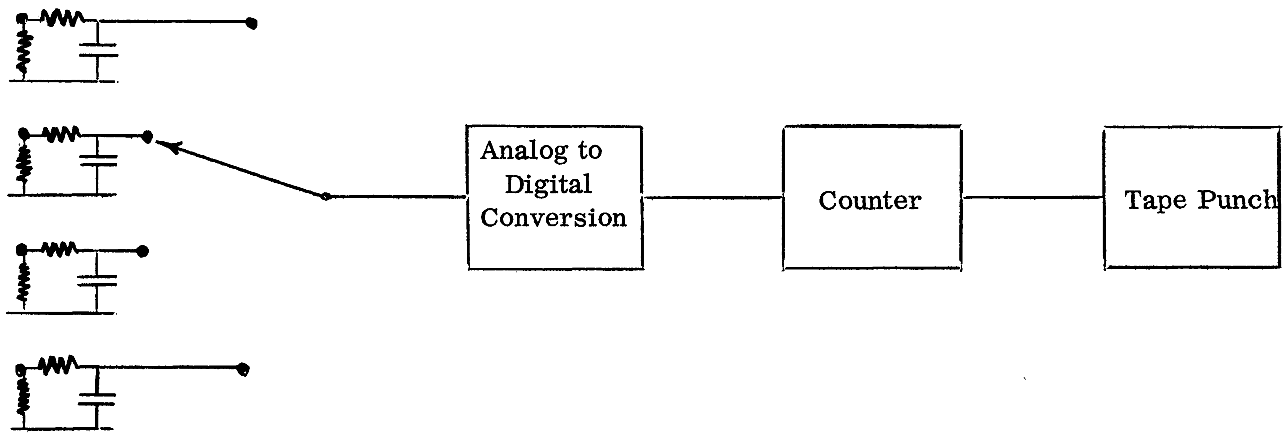

Figure 1

II. REQUIREMENTS FOR THE OUTPUT SYSTEM

1. Rate of sampling a Noise-Free Signal:

The spatial frequency cut-off of the 300-foot telescope at 21 cm is: $f_s = \frac{d}{\lambda} = 430.$

Since the apparent angular velocity of a source at a declination ∂ is:

$$\phi = \phi_o' \cos. \partial$$

With:

$$\phi_o' = 7.3 \cdot 10^{-5} \text{ rad/sec,}$$

the frequency cut-off of the filter equivalent to the antenna is:

$$f_c = f_s \times \phi' = 3.14 \cdot 10^{-2} \cos. \partial \text{ cycle/sec,}$$

and the transit curves must be sampled at intervals less than or equal to:

$$\frac{\Theta}{\text{sec.}} = \frac{1}{2f_c} = \frac{16}{\cos. \partial}$$

For $\partial = 0$, the most unfavorable case, the useful rate of sampling is therefore one point every 16 seconds.

2. Rate of Sampling for a Signal Mixed With Noise:

One can show, (3), that if the signal has a spectrum strictly limited to frequencies between 0 and f_c, and the noise has a spectrum strictly limited to frequencies between 0 and B, all the information contained in a record is specified by discrete values spaced $\frac{1}{B + fc}$. For the ideal case where B = fc, one finds a result identical to the one given by the sampling theorem.

What is important is that in the usual case where B > fc, the necessary rate of sampling is lower than what would have been predicted by the use of the sampling theorem. Actually, it is even not necessary to assume that the noise spectrum is strictly limited and the knowledge of the power frequency response of the filter allows an evaluation of the loss of information corresponding to a given sampling rate. So, in the case of a RC filter, by sampling every time constant one increases the minimum theoretical rms error on the signal by only about 3%; by sampling every half a time constant, this excess error would be reduced to 0.5%, (3).

This means that when we want to use the slowest permissible value of the sampling rate -- and this value Θ_0 is primarily given by the condition that we do not wish to lose any information in the signal: $\Theta_0 = \frac{1}{2fc}$ -- the time constant of the RC filter must be at least:

$$\tau = \frac{1}{2fc} = 16 \text{ sec.}$$

if one agrees to have the rms error on the signal increased by 3%. To reduce the excess error to 0.5% would oblige us to take: $\tau = 32$ sec., which seems to be too high a value for a time constant as far as convenience of use is concerned.

On the other hand, we do not think that the problem of the distortion of the signal due to the time constant, (4), is really of concern since in the restoration of the signal the overall effect of the two filters, radiation pattern and RC filter, can be taken into account.

To sum up, we propose adoption, with a reasonable margin of safety, of a rate of sampling of one point every 10 seconds, and to choose for the RC filters a time constant of 10 seconds. The preceding applies to the most unfavorable case where the declination of the source is: $\partial = 0$, and it is convenient to keep the same rate of sampling for all the transits.

3. Speed of Sampling:

The channel outputs are connected sequentially to the common digitalization system through the stepping switch. Consequently, they are recorded with a delay between them equal to the time necessary for the wiper to switch from one contact to the next. When the number of channels is large, this effect can be significant and has two consequences:

(1) The outputs are recorded at a different moment of the transit curve and their values are therefore different in amplitude. This actually is without importance since by interpolation the final results will be identical.

(2) Difference in time means difference in hour angle and it may be necessary to correct this effect by a corresponding translation of the hour angle scale on the final maps.

The speed of the stepping switch is 70 steps per second. With a 100-channel receiver, the difference in hour angle between the first and the last channel would be 5 seconds of arc, in most cases negligible compared to the 7 minutes of arc of the beamwidth. With a 20-channel, the difference will be only 1 second of arc.

4. Precision of the Measurement:

Since the accuracy of the signal is limited by the noise, the precision of measurement may be reduced accordingly. If σ_o is the rms deviation of the noise, it is sufficient to measure signal plus noise by a number of units such that σ_o is represented by 1 , provided that one agrees to have the total rms deviation increased by 4%, (5). This means that, in this case, with two decimal digits it is possible to measure signals up to 100 σ_o -- or with 8 binary digits signals up to 256 σ_o. Let us consider the case of a 100-channel, 100 kcs bandwidth, receiver -- with a front-end noise temperature of 300°, and low pass filters of the RC type with τ = RC = 10 sec. One has:

$$\sigma_o = \kappa \frac{T}{\sqrt{b\tau}} \doteqdot 1.3 \frac{300}{\sqrt{10^4 \times 10}} \doteqdot 1.2^\circ$$

with 2 decimal digits, S max = 120°

with 8 binary digits, S max = 300°.

For a 20-channel, 100 kcs bandwidth, receiver and with the same assumptions one has

$$\sigma_o \doteqdot 1.3 \frac{300}{\sqrt{10^5 \times 10}} \doteqdot 0.4^\circ$$

Therefore, with 2 decimal digits, S max = 40°

with 8 binary digits, S max = 100°.

We have considered these two possibilities of using either 2 decimal digits or 8 binary digits for they allow us to use only one column of paper tape for recording, which means the possibility of utilizing the tape punch at its maximum speed. Actually, in the case of extragalactic observations at 21 cm, the antenna temperature will not exceed 2 degrees and a much lower precision would be adequate. Even one decimal digit only is enough in this case, since it allows measurements of signals up to 2.8°, taking into account a range of the noise of $\pm 3\ \sigma_o$.

III. TECHNICAL DATA

1. Filters:

Since, whatever the frequency response of a filter, it is possible to calculate accordingly a rate of sampling which allows one to use practically all the available information, and since the distortion of the signal can be taken into account in a general restoration, there is no need to consider questions of optimum filtering. Therefore, the best is to choose the simplest low pass filter, the RC type, which leads, as it has been seen, to convenient values of the time constant and of the rate of sampling. The time constant, $\tau = RC$, is 10 sec. It is desirable in order to avoid too high a value for the capacitors to take R equal to a few megohms, which requires the analog-to-digital converter to have a high input impedance.

With, $C = 5\ \mu F$, $R = 2\ M\Omega$, this input impedance should be of the order of 100 MΩ.

2. Stepping Switch:

To connect the channel outputs to the analog to digital converter, sequentially, either an electromechanical or a solid-state commutator can be used. The main advantage of a mechanical commutator, in addition to low price and provision for switching several independent circuits simultaneously, is the possibility of sampling signals down to the low millivolt levels. On the other hand, the 70 steps per second speed of the Automatic Electric stepping switch used is sufficient in the present case, as it has been seen above.

Actually, a few precautions must be taken in order to make the stepping switch suitable for this commutation of low-level signals. This is due to the fact that the current step function applied to the driving relay coil induces transients in contacts and wires. Measurements made by John Parker, who is taking charge of the complete system, have shown that the effect of this magnetic pick-up can be reduced to a level below the sensitivity of the analog-to-digital converter ($\pm$ 0.3 mv for these measurements) under the following conditions :

--- Presence of the 5 μ F capacitor of the RC filter in the circuit.

--- Delay in recording of 4 ms after the contact has been made.

By use of gold-plated contacts, contact resistance and noise also are reduced. The expected life of this stepping switch is 4 millions of operations, which should mean one year and a half of operation, at the rate of 1 revolution every 10 seconds, assuming a continuous use.

3. <u>Analog to Digital Converter:</u>

The requirements are:

Input impedance : 100 MΩ

Conversion time : 4 ms (taking into account the necessary delay of 4 ms and the contact duration)

Linearity and stability: $\pm$ 0.1%

Resolution adjustable, according to the useful precision wanted from 10 to 250 units full scale.

For convenience of use (measurements of transients, ease in modifying delay of reading, conversion time, input impedance, precision of measurements, etc.) the analog-to-digital converter has been built in the lab. It is based on the principle of slope-time conversion and gives in a gate, whose length is proportional to the applied voltage a train of pulses at a fixed frequency. Its sensitivity is $\pm$ 1 mv.

4. <u>Display:</u>

The train of pulses is converted into the desired code by means of either a binary or a binary-coded decimal counter. In order to store this display during the time of a tape punch cycle -- since this unit cannot be synchrozined -- an "AND NOT" function shift selector is used which shifts the content of the counter and resets it, only after both the counting operation has been finished and a control pulse has been sent by the tape punch. The maximum frequency of the pulses sent to the counter is 65 kcs, and is low enough to allow the use of magnetic core logic elements made by DI/AN Controls, Inc., for the whole display system.

These elements seem to be particularly interesting for all problems concerning digitalization or logic functions, where the frequency is below 100 kcs. Their main advantage lies in their great flexibility which allows one to perform most logic functions by means of a few standard elements externally connected according to basic schemes. Reliability and economy also seem to be characteristic of these elements.

5. Tape Punch:

The Teletype tape punch used is a synchronous unit which perforates paper tape at 110 characters per second, with an 8-level code. The available speed of punching is therefore higher than the speed of sampling (70 steps per second), as far as only one character per channel is enough. (See II,4.) Another advantage of using one character per channel is that it allows us to avoid the intermediate stage of scanning.

For these reasons, we propose to code the digital data in the most convenient way to enter the tape punch (either binary code up to 8 digits, or binary-coded decimal with 2 digits in parallel), and to convert it into the computer code at the level of a tape-to-card converter, for instance. The expected life of such a high speed tape punch is of the order of 3,000 hours, which would mean, with the assumption of a continuous utilization, the need of replacing it every four months. Taking into account the comparatively low price of this unit ($800), this does not seem to be a significant drawback.

6. Accessories:

A channel identification and a digital read-out of the analog-to-digital conversion are provided for. In the case of a 100-channel galactic receiver, it should have been useful also to record sequentially on a one-track analog recorder the channel outputs in order to have an instantaneous, if not accurate, checking of the overall receiver behavior. This does not seem so worthwhile for extragalactic observations since in most cases the signal will be hidden by the receiver noise.

7. Consumption of Paper Tape and Cards:

For a 20-channel receiver, where outputs are sampled every 10 seconds, assuming continuous observation and use of a 2 decimal digits for measurements, the consumption should be 1.5 reel, of 300-foot a reel, of paper tape and 4,800 cards per day (if the tape-to-card conversion is desired).

8. Estimate of Price:

Stepping switch (with power supply)	$120
Tape punch	800
Analog to ditigal converter (with power supply)	350
Channel identification	≃ 50
Digital read-out	≃ 250
Control circuitry for the tape punch and synchronization	≃ 200
Counter and shift selector	≃ 600

Therefore, the price of the complete digitalization set should be below $3,000, exclusive of labor.

For paper tape, cards, and tape punch depreciation, the cost per day of continuous observation is estimated to be $14.

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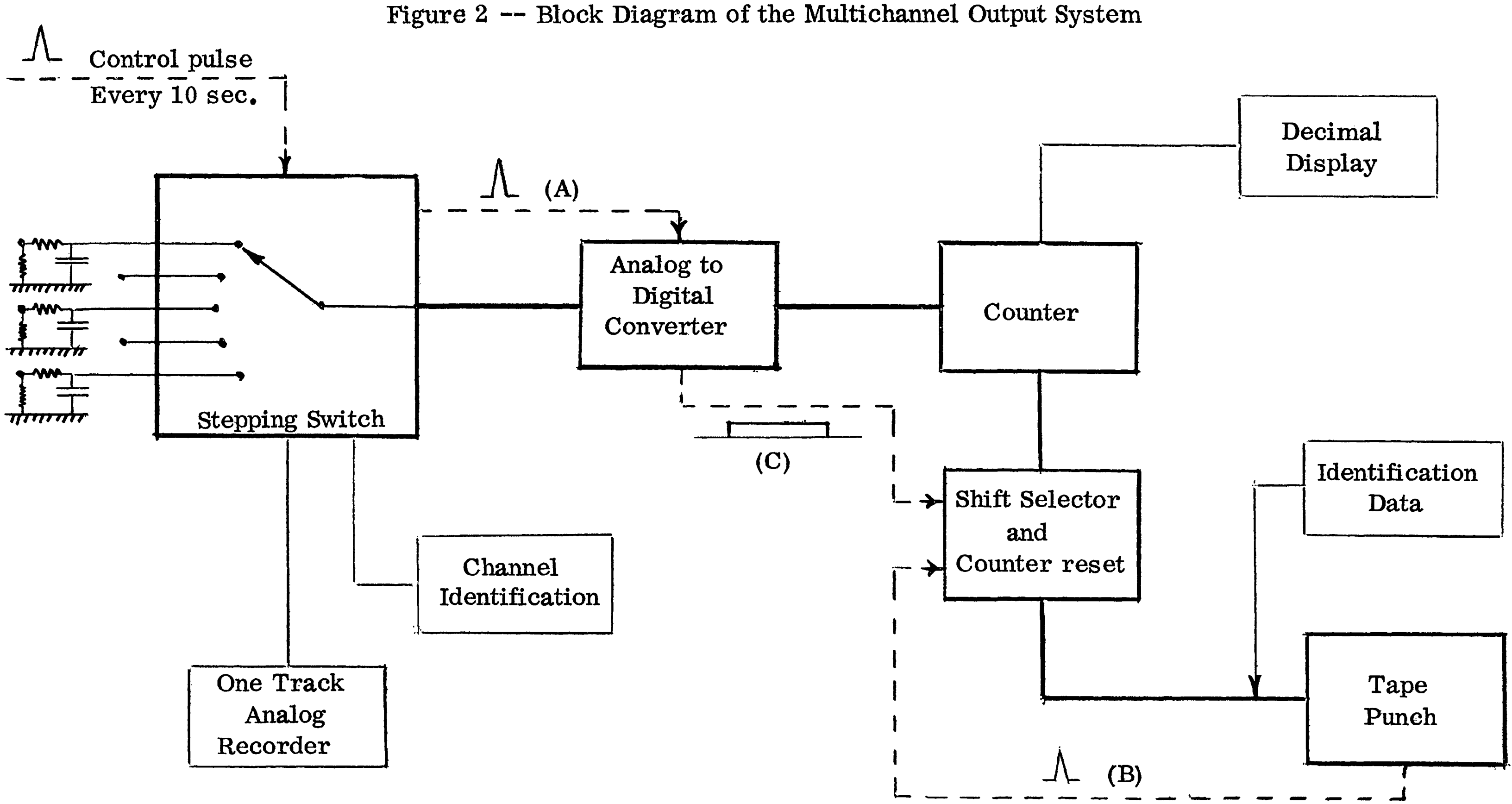
Figure 2 -- Block Diagram of the Multichannel Output System



Timing Diagram