Radio telemetry for upper atmosphere sounding rockets

Pulfer, J.K.

Publisher's version / Version de l'éditeur:
https://doi.org/10.4224/21274414
Report (National Research Council of Canada. Radio and Electrical Engineering Division : ERB), 1963-11

EE

ERB-653

UNCLASSIFIED

NATIONAL RESEARCH COUNCIL OF CANADA

RADIO AND ELECTRICAL ENGINEERING DIVISION

RADIO TELEMETRY FOR UPPER ATMOSPHERE SOUNDING ROCKETS

J. K. PULFER

OTTAWA

NOVEMBER 1963

ABSTRACT

The use of radio telemetry for upper atmosphere sounding is discussed from the experimenter's point of view. Factors affecting the accuracy and information-handling capabilities of telemetry systems are considered. The results of experimental measurements of errors in the standard FM-FM rocket telemetry system used by the National Research Council are given, and the effects of these errors on the design of experiments are discussed. Modifications of the FM-FM system to suit particular user needs, such as high accuracy, wide bandwidth, or the ability to transmit digital data are considered in detail.

CONTENTS

FIGURES

RADIO TELEMETRY FOR UPPER ATMOSPHERE SOUNDING ROCKETS

- J.K. Pulfer -

I. INTRODUCTION

Radio telemetry (literally, "measurement at a distance") is one of the basic tools used in exploring the upper atmosphere. For a large percentage of the experiments in which the measuring equipment is carried aloft by sounding rockets, the most economical way of recovering the resultant data is by means of air-to-ground telemetry. It is important, therefore, for the user to have a thorough understanding of the telemetry systems so that he may design his experiment to utilize their advantages and minimize the effects of their limitations.

From the point of view of the experimenter, the important properties of a telemetry system are the number of channels provided, and the information rate which each channel can handle. The telemetry engineer on the other hand is interested in the power, bandwidth, size, weight, complexity, reliability, and cost necessary to provide a given information-handling capacity.

If the information is in digital form, then the information-handling capability of the channel is easily defined as the number of bits per unit time which can be transmitted at a given error rate. If the information is in analog form, then the channel capacity is determined by the data rate or frequency response of the channel, and the signal-to-noise ratio or accuracy to which a particular analog quantity can be determined.

For sounding rocket experiments, the time available for making measurements of the properties of the upper atmosphere is relatively short compared with that in satellites or deep-space probes. It follows that a high information rate is an essential requirement for rocket telemetry. A second requirement is, of course, versatility. A number of channels and a choice of bandwidths is necessary for all but the most simple experiments. If more than one channel is to be transmitted simultaneously over a single link, then time sharing, frequency sharing, or some other form of multiplexing is necessary.

Some of the telemetry systems in use to-day, with their most important properties and their usual fields of application, will now be described briefly.

1) FM-FM (frequency modulated subcarriers on a frequency modulated transmitter) — The particular subcarrier format which is considered here is that recommended by the IRIG standards [1]. This system offers a series of analog channels with a typical rms error of 1% of the peak-to-peak amplitude range (i.e., 1% of "full scale"). A variety of bandwidths, from a few cycles per second to 2100 cycles per second, are available from the standard system. An advantage of the FM-FM system is the ease with which it may be modified to provide im-

proved accuracy, greater bandwidth capability, or increased operating range. Because it meets most of the engineering requirements as well as the user needs, it has been accepted as the principal system for sounding rockets.

2) Time division multiplexed analog system — Time division multiplex has the advantage of providing a large number of channels each of which has the same basic information capacity. By "super-commutating" or sampling some signals more often than others, multiples of the basic channel width can be obtained, so that the system is very versatile. The chief disadvantages of time multiplex are of an engineering nature, and are brought about by practical limitation on sampling rate, such as size, complexity, reliability, and cost. Time division multiplex is therefore usually limited to low data-rate situations, such as satellite telemetry, deep-space probes, or low data-rate situations in sounding rockets, such as engineering measurements, or equipment monitoring. It is usually used in the form of pulse-amplitude modulation or pulse-duration modulation on a subcarrier in the IRIG FM-FM system in sounding applications. Pulse position modulation amplitude modulating a transmitter (PPM-AM), or pulse amplitude modulation frequency modulating a transmitter (PAM-FM), are sometimes used for engineering measurements in dynamic test firings.

A third form of time division multiplex of analog data is the PFM or pulse frequency modulation used in some satellites. This system is limited to low data rates, but has the advantage that it requires less transmitter power for a given information rate than most other systems of comparable cost and complexity.

3) Pulse code modulation — Analog information can be changed into digital form and transmitted over continuous or discrete channels. Both time division and frequency division multiplex are commonly used. The general term applied to such systems is "pulse code modulation". PCM has several advantages. It can provide increased accuracy in exchange for a decreased data rate. It provides an output in a form which is easily transferred to digital computers for processing. PCM also lends itself to the more complex methods of coding information which reduce the required transmitter power.

As far as upper atmosphere experiments are concerned, PCM is used chiefly to provide a higher accuracy than is available in analog systems. This usually appears in the form of pulse code modulation of one of the subcarriers in the FM-FM link.

An incidental use of PCM is for transmission of data which originates in digital form, such as the output from counting circuits.

More complex coding as a means of reducing transmitter power is usually impractical for sounding rockets, but is used extensively for the low data-rate packages in deep-space probes.

Briefly summarizing the above comparison of systems as applied to upper atmosphere sounding, we can say that at the present time (1963) an FM-FM system is the most desirable from both engineering and scientific points of view. The IRIG standard FM-FM format is used as a basic system.

Accuracy can be improved by using higher than standard subcarrier deviation ratio, by reducing the number of channels, by using multi-channel scale expansion techniques, or by transforming the data into digital form before transmission (PCM-FM-FM).

Data rate can be increased by reducing subcarrier deviation ratio and increasing main transmitter deviation ratio, or by using non-standard, wide deviations.

Versatility in the form of a large number of low data-rate channels can be obtained by time division multiplexing on one or more subcarriers to obtain PAM-FM-FM or PDM-FM-FM operation.

Digital data can be transmitted in either digital or analog form on one or more of the subcarriers.

II. LIMITATIONS OF THE BASIC FM-FM SYSTEM

In the IRIG standard FM-FM telemetry system used in Black Brant rockets, experimental measurements are processed and transmitted to the ground stations as frequency analogs. These analog voltages undergo further processing on the ground before being transcribed as analog or digital paper records. The measured quantity is retained in analog form through a long series of transformations between airborne transducer and final paper record, and at each stage some unavoidable error is added. These errors combine to produce a total value which provides the ultimate limitation on measurement accuracy.

For the purposes of this report, the input to the telemetry system will be considered to be equivalent to the input of one of the subcarrier oscillators. Fig. 1 is a block diagram of one channel of the telemetry system. The output signal from any link in the chain may differ from the input signal because of:

1) distortion due to the nonlinear amplitude and phase characteristics of a single link,

2) additive errors due to channel noise, or the presence of intermodulation products or spurious outputs from other channels,

3) linear filtering due to imperfect amplitude or phase response of a link.

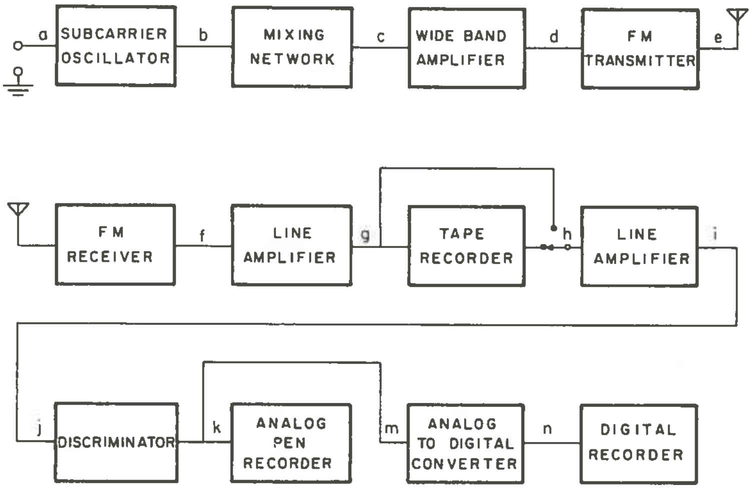

Fig. 1 Block diagram of a single channel of the FM-FM telemetry system

Fig. 2 shows schematically the sources of error in a single telemetry channel. The relative importance of each type of error depends greatly on the signal and how it will be interpreted. If the bandwidth is fully utilized, the linear filtering effect of the channel will produce a greater effect than all other errors combined. It is only in rare cases, however, that a user would consider the filtering as a serious problem since he does not usually transmit information which is completely non-redundant over the full bandwidth. Either a nominally flat amplitude response or linear phase response is adequate in most cases.

The various errors will now be discussed in more detail.

Nonlinear Distortion Within a Single Channel

This type of distortion is produced mainly by nonlinear amplitude characteristics in the subcarrier oscillators and in the discriminators. Specified linearity is $< \pm 0.4\%$ for the SCO's used and $< \pm 0.1\%$ for the discriminators used. This nonlinearity results in an experimentally measured distortion having an rms value approximately 0.3% of full scale, for a sinusoidal signal in the centre of the SCO passband. The use of a pen recorder with a linearity specification of $\pm 1\%$ would approximately triple the distortion. The latter can be avoided by using a post-discriminator analog-to-digital converter and digital records.

Harmonic distortion produced by filtering of the signal while it is in frequency analog form is small in comparison to that occurring to the amplitude analogs.

Fig. 3 is a curve of harmonic distortion for a particular SCO, showing the

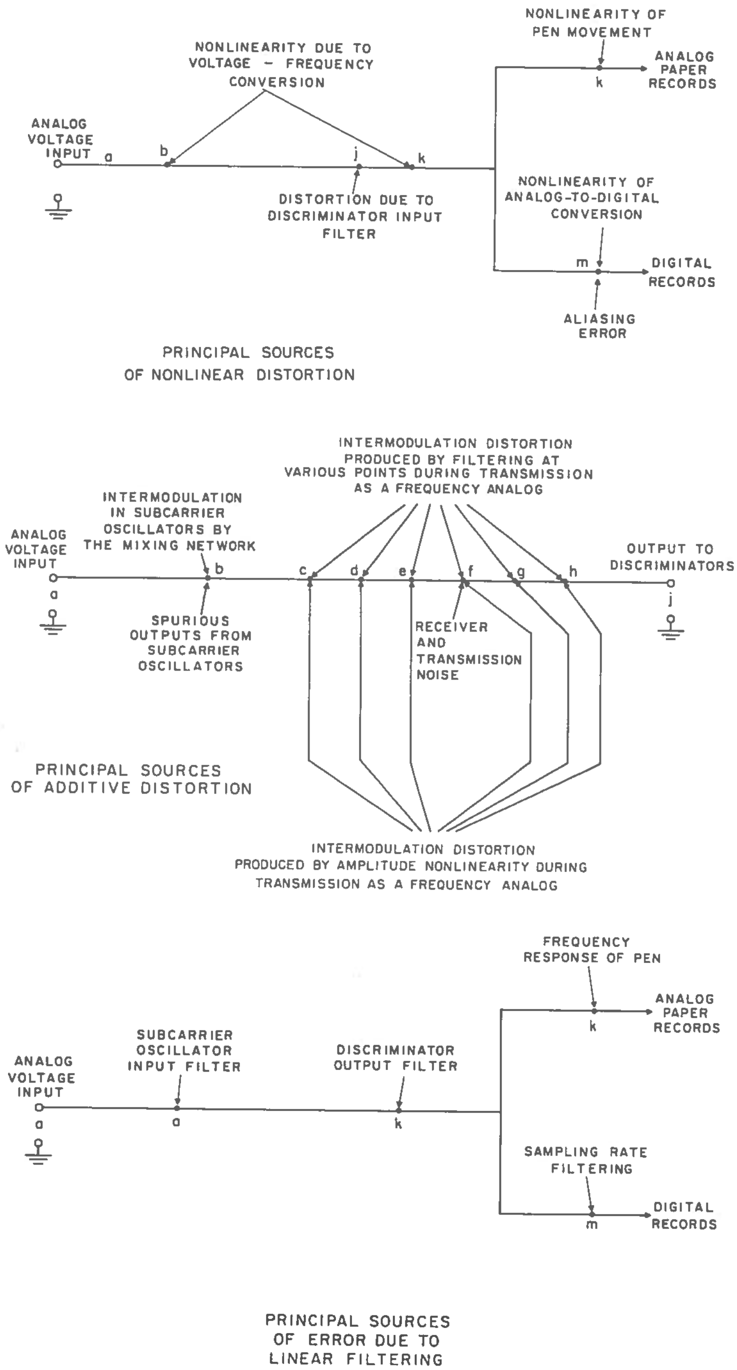

Fig. 2 Line diagrams of a single channel of the telemetry system showing sources and types of error

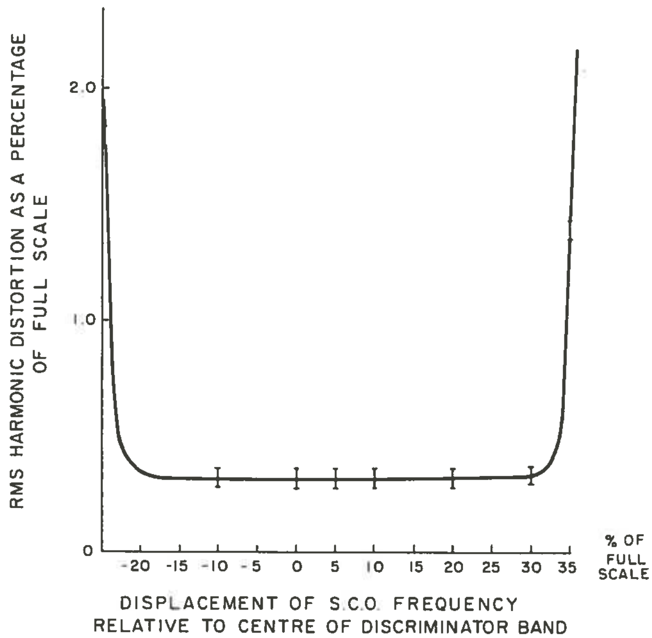

Fig. 3 Harmonic distortion in a particular channel as a function of SCO misalignment

effects of misalignment between an SCO and the discriminator centre frequency. Misalignments of as much as ±10% of full scale have very little effect on system error.

Additive Distortion

Intermodulation distortion is most conveniently stated as peak-to-peak distortion expressed as a percentage of full scale. This result is more useful in most cases because of the relatively high peak-to-rms ratio of the intermodulation noise in an FM-FM system. For our purposes, however, to allow the different forms of distortion to be combined, we will use rms as a percentage of full scale.

Additive distortion occurs at all points in the system, but the only significant contributions occur in the frequency analog portion. Additive noise is added in the transmission link and varies with transmission distance. For the standard IRIG format, and with low noise receiver preamplifiers at 215 to 260 mc/s, a signal of 2 microvolts in a 50-ohm line (approximately -100 dbm) is more than adequate to keep the error due to receiver and cosmic noise less than 0.5%. If proper precautions are taken during the flight, other external sources of noise should be small in comparison with receiver noise.

A second type of additive distortion is the presence in the base band of harmonic outputs from subcarrier oscillators. Although it is difficult to determine the magnitude of the error due to these spurious outputs, it can be as high as 0.3% for the worst combination of SCO frequencies. A more realistic figure which would be encountered when the SCO frequencies are rapidly changing is about 0.07%.

A third source of additive noise is intermodulation distortion. Some IM distortion is produced at every point in the telemetry channel, but the largest contributions occur in the frequency analog parts. The largest contributions are due to the nonlinear modulation characteristic of the transmitter, time delay distortion in the receiver IF amplifier and discriminator, and intermodulation within the SCO's due to signals fed back from the mixing network.

It is quite difficult to determine the magnitude of the various contributions. A measurement of total additive distortion as a function of transmitter deviation helps to point out the large contribution due to the transmitter and receiver. A typical curve for a complete set of randomly modulated subcarriers is shown in Fig. 4.

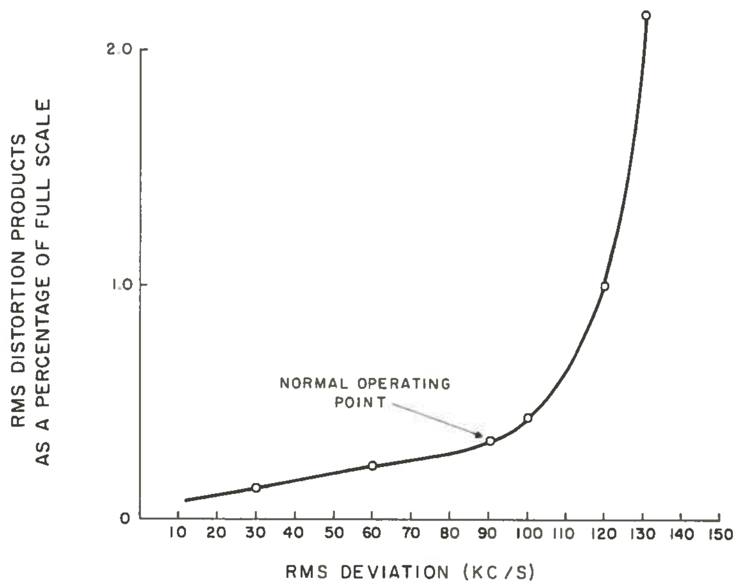

Fig. 4 Total additive distortion in a particular channel as a function of rms transmitter deviation

Linear Filtering

The third principal source of error in the output signal occurs when the output differs from the input in its amplitude and phase versus frequency characteristics because of the various filters encountered. In practice, the principal limitations listed in order of their importance are: the discriminator output filter; the pen response, if paper records are made; or the sampling-rate filtering, if conversion to digital form is used; and the SCO input filter.

The errors produced by linear filtering can be very large compared with other errors when the full signal bandwidth is required. In general, lack of frequency response is accepted as a basic system limitation and is not considered to be a system error, but for those users who are trying to transmit wide-band information, such as vibration, this error is important and must be taken into account. Measurements of filtering error for a typical channel as a function of input filter bandwidth are shown in Fig. 5. The rms error is of the order of 5% when the input data filter has the same bandwidth (3 db down) as the discriminator output filter. Pen response limitations were not included in this measurement because these, in principle at least, can be made arbitrarily small by using the proper galvanometer and damping arrangement.

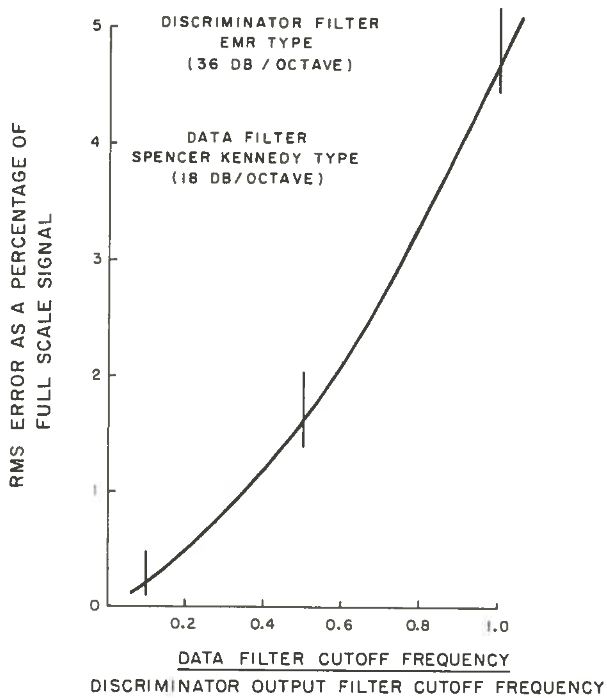

Fig. 5 RMS error introduced by linear filtering in a typical telemetry channel

It should be emphasized that filtering error (like aliasing error in time division multiplex systems) is a function of the filter used to shape the input data, and if a sufficiently severe data filter is used (of the order of 10% of the discriminator filter bandwidth for the IRIG FM-FM system) the error due to linear filtering can be made negligibly small.

MEASUREMENT TECHNIQUES

In order to determine the magnitude of the various contributions to system error, several different experimental setups were used. In addition, where possible, different means were used to determine the same type of error to provide a more thorough understanding of its nature and statistics.

Nonlinear Distortion

The experimental setup for measuring distortion within a single channel is shown in Fig. 6. The distortion meter (Hewlett-Packard Type 330D) uses a sharp rejection filter to remove the fundamental component of the output signal. The meter measures the average rectified value of the remaining components, and is calibrated to read rms value for a sine wave. It therefore tends, in general, to give a reading which is lower than the true rms value. A true rms meter may be connected to the output of the distortion meter to avoid this problem. A second measurement error is introduced since the distortion in the audio source itself is measured by the meter, and this may either increase or decrease the meter reading, depending on its phase relative to the distortion produced by the system.

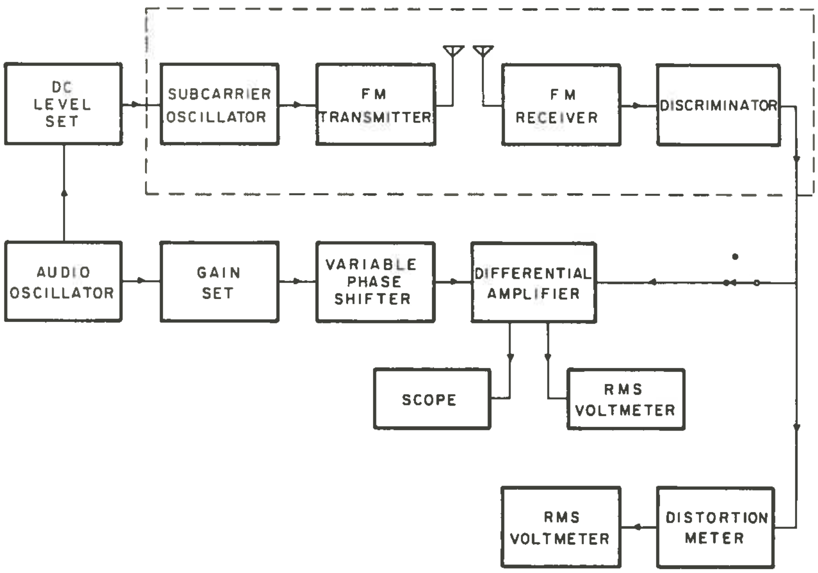

Fig. 6 Experimental setup for measuring harmonic distortion

The second error described above is substantially reduced if the reading is taken from an rms voltmeter driven by a differential amplifier. A further reading can be taken by reading peak-to-peak value of error on the oscilloscope. The only necessary precautions are that the oscilloscope amplifiers are not saturated, and that when the same signal is fed into both channels, the residual output is small compared with system error. Contributions to the output due to additive distortion can be minimized by lowering transmitter deviation, and can be estimated by reducing the sinusoidal input signal to a very small value. Table I gives some typical experimentally measured errors.

TABLE I

SINGLE CHANNEL DISTORTION

Channel 16 (40 kc/s)

Measured IM distortion:	0.09% rms (no signal present)
Measured oscillator spurious output:	0.05% rms
Measured total system distortion:	0.38% rms
Distortion due to non-linearity in channel:	$= [(0.38)^2 - (.09)^2 \pm (.05)^2]^{\frac{1}{2}}$
	$\approx 0.37\%$

Additive Distortion

The experimental setup used to determine additive distortion is shown in Fig. 7. A pseudo-random generator was used to provide input signals for each of the telemetry channels. These signals were fed through appropriate RC filters (18 db/octave) and into their respective SCO's. A set of voltage dividers with values chosen from a table of random numbers was used to provide fixed inputs to the 28 channels of the PAM commutator for the 70 kc/s channel.

The input was removed from the channel being checked and the output was measured by reading the peak-to-peak value on an oscilloscope, as well as the rms value on an rms voltmeter (Fluke Model No. 910A). The contributions due to receiver noise, transmitter distortion, line amplifier distortion, and a tape

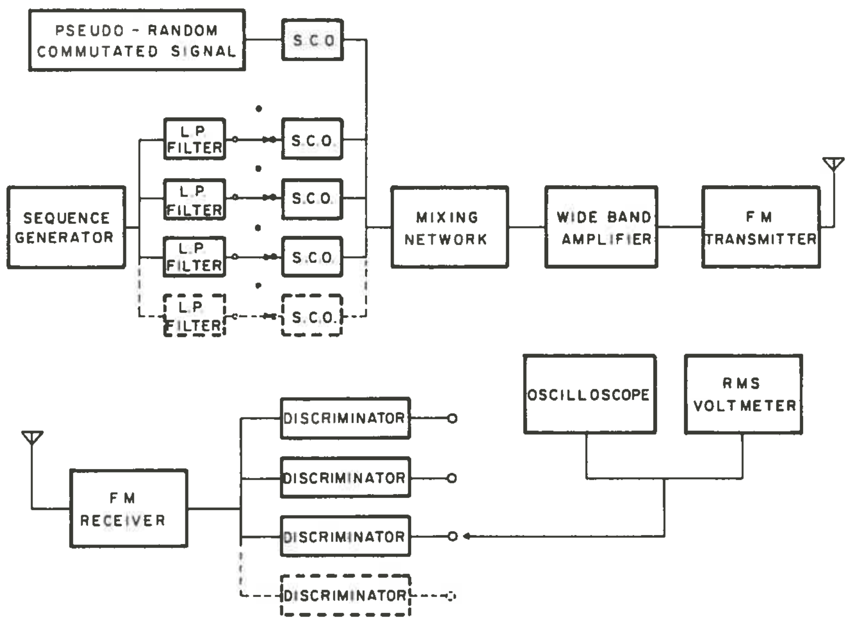

Fig. 7 Experimental setup for measuring additive distortion

recorder could be estimated by varying system parameters. Tables II and III give some typical measured results. More details about the application of pseudo-random signals to distortion measurements are available in a separate publication [2].

TABLE II

ADDITIVE DISTORTION

Channel	Before Tape Recorder	After Tape Recorder
9 (3.9 kc/s)	0.26	0.33
10 (5.4 kc/s)	0.23	0.33
13 (14.5 kc/s)	0.33	0.50
16 (40 kc/s)	0.30	0.50
17 (52.5 kc/s)	0.20	0.40

Figures given are rms distortion as a percentage of full scale. Transmitter deviation was 90 kc/s average.

TABLE III

ADDITIVE DISTORTION

Channel	30 kc/s Deviation	60 kc/s Deviation	90 kc/s Deviation	120 kc/s Deviation
10	0.08	0.13	0.4	1.2
13	0.13	0.23	0.33	1.0
16	0.08	0.13	0.23	0.5

Figures given are rms distortion as a percentage of full scale.
Measurements were made before recording on tape.

Linear Filtering

Measurements of errors produced by linear filtering are the most difficult type to make. It is desirable to be able to compare a signal which has a continuous spectrum with clearly defined properties, with a delayed and filtered version of the same signal at the output of the system. It is also desirable that the shape of the signal spectrum be such that the channel bandwidth can be fully utilized.

Fig. 8 shows the experimental setup used to measure the errors produced by linear filtering. A shift register with feedback is used as a pseudo-random noise source to generate the signal. Outputs are taken from two different stages of the shift register. By varying the bit length, the time delay between the two sequences can be adjusted so that it is equal to the average system delay in a given channel. The two sequences are fed through nearly identical input filters. The delayed filtered sequence and the system output are fed into a differential amplifier and the rms value of the difference is read on an rms meter. Peak readings of the difference can also be made on the oscilloscope screen. In making a measurement for a particular set of conditions, the amplitude of the reference channel, and the bit length of the sequence generator are successively adjusted for a minimum reading on the rms meter. The two filters and the differential amplifier are checked by feeding the same sequence into both and recording the rms value of the residual. Typical values of filtering error results were shown in the curve of Fig. 5.

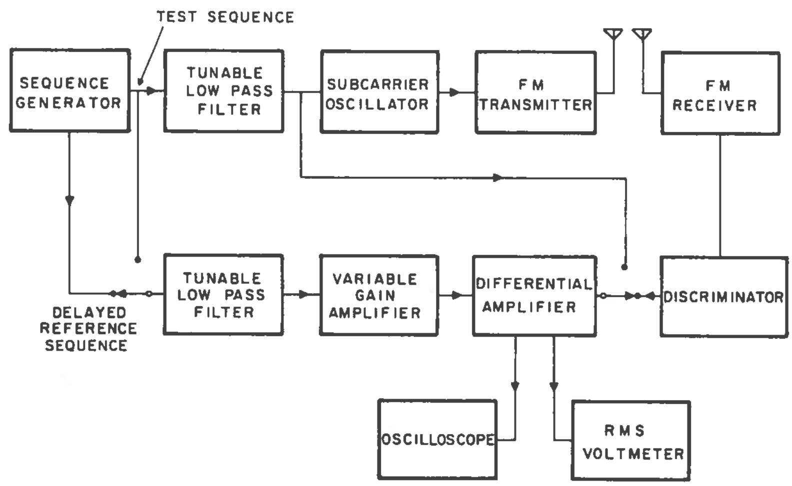

Fig. 8 Experimental setup for measuring errors produced by linear filtering

DISCUSSION

It is evident from the measurements of distortion and the methods of measurement described above, that the accuracy of the telemetry system is not clearly defined, but depends to a great extent on the type of signal which the user desires to transmit.

If we total the errors due to the various sources discussed, we obtain a nominal value of 0.6%, in addition to filtering error. The value of filtering error may be very large or very small. As an extreme example, if the signal changes relatively slowly compared with system bandwidth, then the effects of additive distortion can be integrated out (by using a phase-lock discriminator, for example), and do not contribute significantly under normal conditions. The effects of SCO and discriminator nonlinearity would then be predominant, and would, in most cases, be about 0.3 to 0.5 per cent of full scale. It would be quite possible to "calibrate out" this nonlinearity, however, and reduce the error to perhaps 0.1 to 0.2% of full scale. The remaining error, that due to filtering, would depend entirely on the rate of change of the signal relative to the channel bandwidth. It is thus entirely possible to transmit information over the telemetry system in analog form with a total error of less than 0.5%. The relative merits of trying to use analog format or changing to digital and then transmitting over the same FM-FM channel are discussed in Part III of this report.

As another extreme example, a user concerned with vibration measurements and requiring a bandwidth of several thousand cycles per second would have to use one of the widest channels, and with a deviation ratio less than the standard 5-to-1. Under these conditions, the error could approach 10%.

CONCLUSIONS

It is not possible to specify over-all accuracy for the standard telemetry system. An attempt has been made to describe the system, and its limitations, in such a way that attainable accuracy can be estimated for any given situation.

III. METHODS OF TRANSMITTING ANALOG DATA WITH IMPROVED ACCURACY OVER THE IRIG FM-FM SYSTEM

The problem of transmitting higher accuracy data can be approached in several ways. The first and most obvious approach is to try to improve the components which contribute to the present error. Intermodulation distortion can be decreased by using a transmitter with an improved modulation characteristic, i.e., with improved linearity and frequency response in the modulation circuitry. Improved filtering and increased isolation of the subcarrier oscillators would also contribute to a considerable decrease in the additive components of error. To reduce harmonic distortion, improvements could be made in the linearity of the subcarrier oscillators. Filtering distortion can be controlled by severe pre-filtering of the data, although this may mean some sacrifice in data rate.

If the errors have been reduced by the methods just discussed, new problems arise. Subcarrier frequency drift which, in most systems, is negligible would become noticeable and would have to be reduced. The discriminator response characteristics in both the FM receiver and the subcarrier detectors would have to be improved. Methods of recording and processing the data would also require careful reconsideration.

At this point, other factors must be taken into account. The accuracy of the analog system as a whole can certainly be increased, but only by significantly increasing the cost of both airborne package and ground station. Most users are satisfied with 1 to 2 per cent accuracy, and the increased cost is not justified. It is therefore suggested that, except for improvements in the FM transmitter and subcarrier oscillators, further increases in accuracy should be done on an individual channel basis.

Individual channels can be improved in three ways.

a) Depart from the IRIG standards in one or more channels and increase accuracy by:
   - i) increasing subcarrier amplitude,
   - ii) decreasing number of subcarriers,
   - iii) increasing subcarrier oscillator deviation ratio.

b) Use a scale expansion technique where "fine structure" on the signal waveform is expanded and transmitted by a second channel.

c) Convert the analog data to digital form and transmit as PCM-FM-FM over one or more subcarrier channels.

Departure from the IRIG standards is by far the simplest method of improving accuracy. It will be discussed more fully in Section V.

The scale expansion techniques are illustrated by the commercially available "Vernitel" system. The input amplitude range of zero to full scale is divided up into several equal parts. One channel is used to identify the amplitude section that contains the signal. The amplitude range containing the signal is expanded to full scale value and transmitted over a second channel. Error can be reduced to 0.1% of full scale with careful adjustment, but 0.3 to 0.5% is a more realistic figure. Very large increases in bandwidth or decreases in data rate are necessary to gain increased accuracy by scale expansion techniques. One source [3] estimates that a scale expansion of 16:1, using the Vernitel system, requires a decrease in data rate of 100 to 200 times for continuous random waveforms.

Before comparing the various digital systems, some basic facts about PCM-FM telemetry should be pointed out. In order to obtain practical error rates at the threshold in the FM-FM channels, and to utilize any available bandwidth most efficiently, a bit rate equal to the available channel bandwidth, and a deviation ratio of approximately 0.9 should be used.* This contrasts with a frequency response of one-tenth the available channel bandwidth, and a deviation ratio of five for the standard subcarrier channels.

We must realize, however, that 6 samples-per-cycle and 7 bits-per-sample is a practical minimum requirement for 0.5 per cent accuracy, so that the frequency response of a PCM-FM-FM channel will be approximately one-fifth of that of the corresponding FM-FM channel for the same order of performance. A sacrifice of a further factor of 2, however, to allow 8 samples-per-second and eleven bits-per-sample would provide an accuracy of 0.1%, with allowance for one parity bit and one bit for synchronization. Bandwidths for standard IRIG channels are given in Table IV.

It should be pointed out that the error due to filtering of the PCM channels so obtained would be of the same order as that in corresponding FM-FM channels, and severe pre-sampling filtering is necessary if continuous spectra, such as vibration data, are to be transmitted. At the present time (1963) IRIG standards do not exist for PCM-FM-FM.

---

* See Part IV.

TABLE IV

DATA BANDWIDTH FOR TRANSMISSION OF ANALOG INFORMATION IN CONTINUOUS OR DISCRETE FORM

IRIG Channel	FM-FM System with 1% Error (cps)	PCM-FM-FM System with 0.1% Error (cps)
E	2100	230
D	1600	180
C	1200	130
B	900	100
A	660	73
18	1050	115
17	790	88
16	600	67
15	450	50
14	330	37
13	220	24

All of the above reasoning has been based on the idea that the information would be changed from analog to serial binary digital form, and transmitted over one subcarrier oscillator. Other possible formats for the digital information are:

1) All Parallel Binary — one channel per digit

2) Serial Binary Coded Decimal

3) Series Parallel Binary Coded Decimal, i.e., one decimal digit per channel in serial form

4) All Parallel Binary Coded Decimal, i.e., 4 channels per decimal digit

5) Staircase Parallel Binary, i.e., several binary digits per channel added into staircase form

6) Staircase Binary Coded Decimal, i.e., one decimal digit per channel in staircase form

These possible formats show the extreme flexibility of handling the information once it is in digital form. Bandwidth per channel, number of channels, and the accuracy which each channel must provide can be interchanged at will. The most serious limitation on the flexibility is the IRIG standard for the FM-FM link. The standard FM-FM system uses constant percentage bandwidth for each subcarrier channel, so that in any application where information is to be sent in parallel form over a number of channels, the different frequency response and delay of each channel must be considered and, if necessary, compensated for. If only three parallel channels were required, identical input and output filters on three adjacent IRIG channels could be used. The subcarrier amplitude could be adjusted to make the S/N in each channel identical. (In fact, a linear taper of SCO amplitudes has other advantages in reducing IM distortion.)

The bandwidth requirements for the various digital formats can be outlined qualitatively as follows:

1) Parallel channels require no more bandwidth, in principle, than serial operation, and provide some advantages in data processing on the ground.

2) BCD channels may require up to 25% more bandwidth than straight binary, depending on the system accuracy.

3) Staircase waveforms (or multi-level digital systems) offer increased data rates at the expense of an increased error rate. They are intermediate between analog FM-FM and binary PCM-FM-FM.

CONCLUSIONS

Briefly there are four approaches to improved accuracy within the IRIG FM-FM system.

1) Increased accuracy may be obtained with the standard analog system, by improving individual components. It appears that an over-all accuracy of better than 0.5% is not practical by this approach because of the large expense involved in obtaining better equipment and keeping it in peak operating condition.

2) Increased accuracy may be obtained by using analog scale expansion techniques. This system involves less cost and expenditure of effort, but data rates must be reduced by at least a factor of 100 in comparison with simple FM-FM in order to achieve 0.2 to 0.3% accuracy.

3) Increased accuracy may be obtained within the analog system by exchanging data rate for accuracy. Increased deviation ratio on a given subcarrier channel by reducing the cutoff frequency of the data filter is the most obvious example.

4) The last approach is to convert the data to digital form in the rocket and transmit by some form of PCM-FM-FM. The format used would depend on the relative amounts of processing equipment desirable in the rocket and on the ground, the acceptable error rate, and the nature of the signal being measured.

IV. TRANSMISSION OF DIGITAL DATA OVER THE FM-FM ANALOG TELEMETRY SYSTEM

In Part III, the problems of transmitting analog data in digital form over a conventional FM-FM channel are discussed. The results may equally well be applied to the direct transmission of binary or multi-level digital data. Since channel 18 is usually used for PAM-FM-FM, the widest band which can readily be made available for binary data is the high deviation 40 kc/s band or channel C.

Let us consider the problem of transmitting binary information by frequency modulating the 40 kc/s subcarrier. The properties of this channel are a nominal bandwidth of 12 kc/s and a nominal signal-to-noise ratio of 15 db.

When used as an analog channel, with a deviation ratio of 5:1, the data bandwidth is reduced to 1200 cps, and the signal-to-noise ratio is improved to approximately 40 db. It is extremely wasteful to transmit binary data with the same deviation ratio as used for analog data, since the signal-to-noise ratio of 40 db is much greater than that required to determine which of the two amplitude levels has been transmitted.

An experimental investigation of a PCM-FM-FM transmission system was conducted to determine the optimum values of parameters, such as bit rate and deviation ratio.

The results of the investigation are briefly as follows. The optimum value of bit rate is the maximum which can be transmitted without excessive amplitude reduction of single bits; i.e., the optimum bit rate is equal to the channel bandwidth (12,000 bits/second for channel C). Once the bit rate has been chosen, the deviation ratio for smallest error rate depends on the statistics of the digital sequence being transmitted and the type of detector used. For a sampling type detector, and a pseudo-random, non-return-to-zero (NRZ) sequence, the optimum value of peak deviation ratio is approximately 0.9:1. Bit error probability (P_ϵ) with a signal-to-noise ratio of 15 db and optimum deviation ratio is approximately 5×10^{-8}. For channel C with 12,000 bits/second, this corresponds to an average time between errors of about 30 minutes.

Since this error rate is still unrealistically low for a 6-minute sounding rocket flight, a still higher number of bits per second can be used. The experimentally measured error probability for a data rate of 15,000 bits per second was 10^{-5} or about one error every 7 seconds.

Table V gives the values of binary data rates over each of the FM-FM subcarrier channels for the two conditions described above.

TABLE V

TRANSMISSION OF BINARY DATA OVER THE IRIG FM-FM SYSTEM

Channel	Nominal Analog Bandwidth (cps)	Optimum Binary Data Rate $P_\epsilon = 5 \times 10^{-8}$	Binary Data Rate For $P_\epsilon = 10^{-5}$
E	2,100	21,000	26,000
D	1,600	16,000	20,000
C	1,200	12,000	15,000
B	900	9,000	11,000
A	660	6,600	8,250
18	1,050	10,500	13,000
17	790	7,900	10,000
16	600	6,000	7,500
15	450	4,500	5,600
14	330	3,300	4,100
13	220	2,200	2,700
12	160	1,600	2,000
11	110	1,100	1,370
10	81	810	1,010
9	59	590	740

If it is desirable to send multi-level digital pulses at the same error rate, the deviation ratio of the subcarrier oscillator must be increased to provide a better signal-to-noise ratio. This, in turn, means that the repetition rate of the pulses must be decreased. To resolve 3 levels with an error rate of better than 10^{-5} would require a pulse rate of something less than 8000 per second. A greater rate could, of course, be used if a higher number of errors could be tolerated.

V. MODIFICATIONS OF THE BASIC IRIG FORMAT TO INCREASE OPERATING RANGE OR DATA RATE

As has been discussed in Part III above, it is possible to modify the IRIG format in many ways which will allow certain transmission parameters to be improved at the expense of others. No component modifications are necessary for these format changes, so that no additional expense or complexity is involved.

Two types of compromises are possible:

a) Some channels can be improved at the expense of others by altering the base-band format. The improvement can be in the form of accuracy, operating range, data rate, or combinations of these.

b) Some parameters of one or more channels can be improved at the expense of other parameters in the same channels.

Let us first consider modifications of type (a). Some possibilities are:

i) Increase the amplitude of some subcarriers while decreasing that of others to maintain the same over-all transmitter deviation. This will provide an increase in signal-to-noise ratio in some channels and a decrease in others.

ii) Decrease the number of subcarriers, and increase the amplitudes of each so as to maintain the same over-all transmitter deviation.

iii) Decrease the number of subcarriers and decrease the receiver IF bandpass. (Most telemetry receivers in use today have plug-in or switchable bandwidths.) Transmitter deviation is allowed to decrease by keeping the remaining subcarrier amplitudes constant.

iv) Replace some of the lower subcarriers by a band of information directly modulated on the transmitter.

v) Remove some of the subcarriers and increase the deviation of the remaining ones (e.g., use channels A, C, and E, in place of 13, 14, 15, 16, 17, and 18).

The improved signal-to-noise ratio obtained by the above modifications can be used directly to decrease the error due to additive distortion at a given distance or as a means of increasing the operating distance at a given error.

The improved signal-to-noise ratio may be sacrificed for higher data rate or a decrease in error due to filtering in a given channel by modifications of type (b) as suggested below:

i) Retain the deviation ratio in a given channel, but increase the bandwidth of the discriminator output filter. This will decrease the errors due to linear filtering, at the expense of increased additive distortion.

ii) Increase the deviation ratio in a given channel by decreasing the bandwidth of both the input data filter and discriminator output filter. This will reduce errors due to additive distortion.

iii) Decrease the deviation ratio by increasing the bandwidth of both the data filter and the discriminator output filter, to allow an increased frequency response in exchange for larger errors.

It is, of course, possible to do modifications of types (a) and (b) either separately or together, depending on the particular system characteristics which are desired.

Some examples which illustrate the flexibility of the FM-FM system are given below:

1) Increased data rate in exchange for accuracy: Band E subcarrier can be modulated with data from 0 to 10,500 cps with a reduction in signal-to-noise ratio of approximately 15 db.

2) If the user is willing to sacrifice linearity and does not require information below about 100 cps, the main telemetry transmitter can be modulated directly with a bandwidth from 100 cps to 20 kc/s. This represents a 10- to 20-fold increase in bandwidth at no sacrifice in signal-to-noise ratio.

3) Two channels may be used — one, a wide-band direct-modulated channel 100 cps to 10 kc/s with deviation ±90 kc/s, the other, a 14.5 kc/s subcarrier oscillator which will provide DC - 220 cps at high quality by deviating the transmitter ±50 kc/s. Range can be increased by a factor of two over the normal operating distance.

4) Narrow-band low-power package: Two subcarrier oscillators (channels 12 and 13), each with a deviation ratio of ±30 kc/s. The IF bandpass of the receiver is reduced to 100 kc/s. The transmitter power required is 10 db below that of the standard system for the same operating range.

VI. CONCLUSIONS

In contrast to popular belief, the FM-FM system now used in upper atmosphere sounding is extremely flexible. High accuracy, wide bandwidths, and increased operating range are properties which can be obtained without component modifications or purchase of additional equipment. It must be remembered, however, that the information capacity of the system as a whole is limited, and an improvement in any particular parameter means a sacrifice at some other point. It is also important to realize that the relationships between parameters are, in general, not linear, so that large sacrifices may be necessary in some, to secure small gains in others.

VII. ACKNOWLEDGEMENT

The author would like to acknowledge the contributions of Mr. T.H. Shepertycki to the measurements of telemetry system errors, and of Mr. F.V. Cairns in many helpful discussions during the preparation of this report.

VIII. REFERENCES

1. IRIG Document No. 106-60, Armed Services Technical Information Agency, Arlington, Va., June 1962

2. T.H. Shepertycki, "Telemetry Error Measurements Using Pseudo-Random Noise" (to be published)

3. F.M. Riddle and J.F. Meyer, "The Vernitel System", Internal Memo, Jet Propulsion Laboratory, Cal. Inst. Tech., January 1960