

Final Report

SIMULATION STUDIES FOR NOAA STABILIZED COMPENSATION PROGRAM

Prepared under Contract No. 2-35369

by

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#### FOREWORD

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The program studies began on 21 April, 1972, and were completed on 20 May 1973. Participating RTI staff members were as follows:

- W. H. Ruedger, Project Leader
- G. S. Brown, Member of the Technical Staff
- J. H. White, Member of the Technical Staff



#### ABSTRACT

<span id="page-4-0"></span>This report describes the participation of the Research Triangle Institute in the NOAA/NESS program to achieve data quality assurance. Long-term observation of the quality of the data from the scanning radiometer systems aboard the ITOS type spacecraft has indicated that a requirement exists for a systematic means of maintaining data quality.. Previous studies have further indicated that the alignment of the equipment used to compensate for spacecraft tape recorder flutter effects was particularly sensitive in affecting data quality. This equipment was selected as the subject of a pilot program to assure data quality by stabilized control. RTI has contributed to the overall objective of this pilot program by conducting analyses of the z-axis\* compensation technique as it interacts with the data and by recommending diagnostic logic to survey data quality as the data are ingested from the Command and Data Acquisition Station (CDA).

\*NOAA nomenclature referring to the flutter components introduced at the spacecraft tape recorder.



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#### 1.0 INTRODUCTION

<span id="page-11-0"></span>This report describes the participation of the Research Triangle Institute in the NOAA/NESS program to achieve data quality assurance. Long-term observation of the quality of the data from the scanning radiometer systems aboard the ITOS type spacecraft has indicated that a requirement exists for a systematic means of maintaining data quality. Previous studies have further indicated that the alignment of the equipment used to compensate for spacecraft tape recorder flutter effects was particularly sensitive in affecting data quality. This equipment was selected as the subject of a pilot program to assure data quality by stabilized control. RTI has contributed to the overall objective of this pilot program by conducting analyses of the z-axis compensation technique as it interacts with the data and by recommending diagnostic logic to survey data quality as the data are ingested from the Command and Data Acquisition Station.

Figure 1-1 shows the overall flow of ITOS scanning radiometer data. During real time operation, the data are recorded on the spacecraft tape recorder for subsequent readout. During acquisition the tape recorder is played back at an increased speed, the data is frequency division multiplexed, and then transmitted to the ground station via an S-Band telemetry link. At the ground station the data is received, demultiplexed and recorded. During recorder playback, the data is z-axis corrected and fed to the long lines for transmittal to DAPAD where it is digitized and input to the data processing equipment.

Figure 1-2 indicates the overall philosophy of the program to stabilize data quality. The concept consists of monitoring specific parameters in the data as they are ingested into the Digital Data Handling System (DDHS) , conducting analyses in real-time, evaluating these analyses and communicating informative diagnostics to the CDA if data quality is not being maintained. In the event a fault does exist at the CDA, it is anticipated that these diagnostics will be of assistance in localizing and correcting this condition. This program represents an on-going effort at NESS. As a result, this report documents only those areas in which RTI was directly involved. This involvement consisted of an investigation of the compensation technique, the development of a digital model of the hardware, and the use of this model to derive sensitivities to be used as guidelines in the development of the required diagnostic logic.

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Figure 1-2. Overall Data Quality Assurance Philosophy.

The report has three main features. The initial sections present a tutorial discussion of frequency modulation theory to serve as background and also a tutorial discussion of the theory of operation of the z-axis compensation technique. Subsequent sections describe the model, pertinent analyses conducted with the model, and preliminary diagnostic logic recommendations based on these analyses. The computer algorithm is then documented in detail in the appendices. It is hoped that this approach will result in the report being a working document for NESS and RTI personnel as well as documenting the effort expended during the study.



#### 2.0 FREQUENCY MODULATION CONSIDERATIONS

<span id="page-15-0"></span>This section presents a tutorial introduction to the mathematical aspects of frequency modulation. The intent here is to indicate the effects of the modulation technique being non-linear. Of prime concern is the spectral representation of an FM modulated waveform, especially the facts that spectral superposition does not apply and that tone modulation has a distributed spectral representation which is tone amplitude dependent.

## 2.1 Theoretical Considerations

The "frequency" of a waveform is defined as the time-rate-of-change of phase and as such usually suggests a periodic phenomenon (i.e. in the spectral sense). On the other hand "frequency modulation" implies a time varying frequency which is contrary to the usual notion. It therefore becomes important in the context of frequency modulation to distinguish between "spectral" and "instantaneous" frequencies.

In general, a frequency modulated wave  $x^{\dagger}_0(t)$  can be written as

$$
x_c(t) = A_c \cos\theta_c(t) \tag{2-1}
$$

where A<sub>c</sub> is constant and  $\Theta_c(t)$  is a function of the carrier frequency and some baseband signal  $x(t)$ . In order to distinguish between the two types of frequency mentioned in the preceding paragraph,  $\Theta_{c}(t)$  can be written as

$$
\Theta_{\rm c}(t) = 2\pi f_{\rm c}t + \phi(t)
$$

where f is the carrier (spectral) frequency and  $\phi(t)$  contains the baseband signal  $x(t)$ . Returning to the concept that frequency is the time-rate-of-change of phase, one can define "instantaneous" frequency as

$$
\xi(t) = \frac{1}{2\pi} \frac{d}{dt} \theta_c(t) = f_c + \frac{1}{2\pi} \frac{d\phi(t)}{dt}
$$

If the baseband signal is now taken to be the change in instantaneous frequency as a function of time,

$$
\xi(t) = f_c + f_p x(t)
$$

<span id="page-16-0"></span>(where  $f_n$  is called the "deviation" constant and is a system parameter), one can identify

$$
\frac{d\phi(t)}{dt} = 2\pi f_D x(t)
$$

or

$$
\phi(t) = 2\pi f_D \int_0^t x(\tau) d\tau .
$$

Now the modulated wave given in eq. 2-1 can be written as

$$
x_c(t) = A_c \cos[\omega_c t + 2\pi f_D \int_0^t x(\tau) d\tau]
$$
 (2-2)

indicating the relation between a baseband signal and the modulated carrier.

The point of this discussion is to demonstrate that an intuitive notion of the relationship of the spectral distribution of signal information at baseband and its distribution after modulation is not available. This will be pursued in greater detail in the following section which deals with the spectrum of a frequency modulated wave.

#### 2.2 Spectral Analysis of Modulated Carrier

The spectral analysis of a frequency modulated carrier is pertinent to the study in that data processing equipment bandwidths are continually of interest. For this reason a discussion of the behavior of the spectral distribution of the modulated carrier is included. The approach will be to initially examine the behavior of a carrier modulated by a single tone as a function of tone amplitude and frequency and to then extend this to a two tone situation. Multiple tone situations will not be discussed as they are simply mathematically complicated extensions of the two tone case and offer no additional insight.

Consider the baseband tone,

 $x(t) = A_m \cos \omega_m t$ 

Equation 2-2, the modulated carrier takes on the form,

$$
x_c(t) = A_c \cos(\omega_c t + 2\pi f_D \int_0^t A_m \cos \omega_m \tau d\tau)
$$

$$
= A_c \cos(\omega_c t + \frac{f_D A_m}{f_m} \sin \omega_m t) .
$$

Define,

$$
\beta\ =\ \frac{f_D A_m}{f_m}
$$

and obtain

$$
x_c(t) = A_c \cos(\omega_c t + \beta \sin(\omega_m t)) \qquad . \tag{2-3}
$$

The parameter  $\beta$  is termed the modulation index and represents the maximum phase deviation produced by the tone. Rewriting equation 2-3 as

$$
x_c(t) = A_c[\cos\omega_c t \cos(\beta \sin\omega_m t) - \sin\omega_c t \sin(\beta \sin\omega_m t)]
$$

and  $\textsf{writing } \cos(\beta \textsf{sin} \omega_{\textsf{m}} \textsf{t})$  and  $\sin(\beta \textsf{sin} \omega_{\textsf{m}} \textsf{t})$  in terms of Bessel functions of the first kind, one can achieve a Fourier series type representation for  $x_c(t)$ . In particular,

$$
x_{c}(t) = A_{c}J_{o}(\beta) \cos \omega_{c}t
$$
  
+ 
$$
\sum_{n \text{ odd}}^{\infty} A_{c}J_{n}(\beta) [\cos (\omega_{c} + n\omega_{m})t - \cos (\omega_{c} - n\omega_{m})t]
$$
  
+ 
$$
\sum_{n \text{ even}}^{\infty} A_{c}J_{n}(\beta) [\cos (\omega_{c} + n\omega_{m})t + \cos (\omega_{c} - n\omega_{m})t].
$$

Notice that the spectrum of the modulated carrier consists of an infinite number of lines spaced at intervals of  $\omega_{_{\rm I\!R}}$  about the carrier with amplitudes that vary as Bessel functions of the modulation index. An illustrative line <span id="page-18-0"></span>spectrum for a frequency modulated tone is shown as Figure 2-1.



Figure 2-1. An FM line spectrum, single tone modulation.

Further, notice that for a given tone frequency, the modulation index 3 changes as the amplitude of the tone varies which produces a different spectral distribution. Figure 2-2 shows the behavior of Bessel functions versus their argument.



Figure 2-2. Bessel functions of fixed order plotted versus the argument  $\beta$ .

One may view the variation in spectral distribution as a function of amplitude and frequency by noting that the harmonics generated by  $\beta=2$  are quite different from harmonics generated by  $\beta=4$ . This differential in  $\beta$  could be simply the result of a two-to-one increase in tone amplitude. Notice that for  $\beta=12$ (which incidentially represents an unlikely situation), the maximum spectral line occurs at n=10 which represents a frequency ten times the tone frequency and would require an extremely wide channel bandwidth. Figure 2-3

<span id="page-19-0"></span>indicates the effect of changing tone amplitude and frequency. The effect of amplitude only and frequency only changes are indicated (recall that  $R_{\rm 0}$ ,  $\frac{A_{\rm m}}{M}$ 

m



Figure 2-3. Tone-modulated FM line spectra showing the effects of tone amplitude and frequency. (a) frequency fixed, increasing amplitude; (b) amplitude fixed, decreasing frequency.

The preceding discussion has treated the behavior of the spectrum for only a single tone. It is of interest to consider the additional influence of a multiple tone signal as this represents a more realistic situation. The mathematics of investigating the general multitone situation is extremely cumbersome and, as a result, only the two tone case will be discussed here. This is adequate as all the pertinent effects will be revealed.

Following the mathematical procedure introduced for a single tone and omitting the intermediate details, for a signal of the form

 $x(t) = A_1 \cos\omega_1 t + A_2 \cos\omega_2 t$ 

where  $f_1$  and  $f_2$  are not harmonically related, one can write the modulated carrier as,

<span id="page-20-0"></span>

**Figure 2-4. Spectra Showing the Failure of Superposition in Frequency Modulation.**

$$
x_c(t) = A_c \sum_{n=1}^{\infty} \sum_{m=1}^{\infty} J_n(\beta_1) J_m(\beta_2) \cos(\omega_c + n\omega_1 + n\omega_2) t
$$

Again, without including the details, this may be interpreted in the spectral domain as being divided into four categories:

1. the carrier alone at  $f_{\rm c}^{\rm c}$ 

- 2. sidebands at  $f_c \pm nf_1$  due to  $f_1$  alone
- 3. sidebands at  $f_c \pm mf_2$  due to  $f_2$  alone
- and 4. sidebands at  $f^ + n f^1 \pm m f^2$  due to the beat-frequency modulation between the two tones.

This last category is the result of the modulation process being non-linear and, as would be expected, gives rise to the feature of FM that spectral superposition does not hold and as such data interaction can occur if the modulated carrier is bandlimited to any extent. Figure 2-4 demonstrates this effect by presenting the single and sum spectra of two tones.

The preceding discussion has been brief, but has demonstrated the data dependence of the modulated subcarrier spectrum. The basic reason for including this discussion is to admit that bandpass filtering the modulated carrier may cause distortion of the baseband signal and interaction of spectral harmonics. Standard rules of thumb are common in selecting bandwidth requirements and are conditioned on the data characteristics. For example "Carson's Rule" provides that for most applications, a bandwidth of  $\pm$  (f<sub>D</sub> + f<sub>m</sub>) about the carrier frequency is adequate  $(f_{\text{D}}$  is the previously mentioned deviation constant and  $f_m$  is the highest frequency in the baseband signal). These topics are explored in greater detail in references 2-1 and 2-2.



#### 3.0 Z-AXIS RACK OPERATION

<span id="page-23-0"></span>This section presents a qualitative discussion of the operation of the z-axis rack. Included is an introduction to pulse-averaging discriminator operation and utilization of the output of the reference discriminator in order to compensate for tape recorder speed error.

#### 3.1 Discriminator Operation

A block diagram of the basic pulse-averaging discriminator (see ref. 3-1) used in the z-axis rack is shown in Figure 3-1. The input bandpass filter shown is not necessary for the discussion but is included for completeness.



Figure 3-1. Basic Discriminator Block Diagram.

Figure 3-2 indicates waveforms encountered at various locations in the discriminator and may be used to describe the functional operation of this type of FM demodulator. Figure 3-2-a indicates a simple 290 Hz base-band tone which will be used to modulate a 5000 Hz sub-carrier. (It should be noted that the waveform parameters such as subcarrier frequency, etc., were chosen for ease of graphic presentation rather than as representative of the actual hardware configuration.) A portion of the modulated subcarrier is shown in Figure 3-2-b. This represents the input to the discriminator.

This frequency modulated subcarrier is first "hard limited" at the input in order to remove any residual amplitude modulation that may have been introduced in the satellite modulator, multiplexer and transmitter or in the ground station receiver and demultiplexer. The resultant waveform is shown in Figure 3-2-c. It should be remarked that limiting does not theoretically introduce base-band distortion in that all base-band information

<span id="page-24-0"></span>

Figure 3-2. Basic Discriminator Operation - Waveforms.

<span id="page-25-0"></span>is contained in the zero-crossings of the frequency modulated subcarrier. These crossings are accurately preserved in the process of limiting.

The limited subcarrier is then fed to a Schmitt Trigger for purposes of detecting the aforementioned zero crossings. The Schmitt Trigger outputs a pulse each time the limited waveform passes through zero. A separate pulse is output depending on whether the zero crossing was positive going or negative going. The upcrossings are indicated in Figure 3-2-d and the downcrossings in Figure 3-2-e. The upcrossings are fed to detector "A" and the downcrossings to detector "B" where the pulses are "stretched" to have a width which is one-fourth the period of the unmodulated subcarrier. This choice of pulse width is not arbitrary. It results from the constraints that an unmodulated subcarrier will provide zero at the output of the demodulator and that one of the effects of the "NOR" is to act as a frequency doubler. Figure 3-3 indicates the relationship between the subcarrier period and pulse width in the unmodulated situation. (This figure will be more informative if referred to again following the discussion of the "NOR" circuit and low pass output filter.)



T is period of the subcarrier

 $\tau$  is the detector pulse width (=  $T/4$ )

Figure 3-3. Constraints on Detector Pulse Width.

<span id="page-26-0"></span>

Figure 3-4. Z-Axis Rack Compensation Technique. Figure 3-4. Z-Axis Rack Compensation Technique.

<span id="page-27-0"></span>The outputs of detectors "A" and "B" are shown in Figures 3-2-f and 3-2-g, respectively.

The stretched pulses are fed to the "NOR" circuit. This circuit produces an "ON" condition if there is a pulse from neither detector "A" nor detector "B" as shown in Figure 3-2-h. Notice that the time the waveform spends in the "ON" condition varies with time while the time spent in the "OFF" condition does not. This waveform when low-pass filtered (i.e. averaged) will produce the waveform shown in Figure 3-2-i. Notice that the original base-band signal is recovered exactly except for a change in polarity. The polarity is reversed in the output amplifier.

#### 3.2 Flutter Compensation Technique

The effect of tape speed error on the spacecraft tape recorder is to shift the data sub-carrier frequency to a higher frequency when the recorder is running faster than nominal and to a lower frequency when the tape recorder is running slower than nominal. Since flutter is a time varying error in recorder speed, its effect is to deviate the carrier from nominal in some time varying fashion. This is equivalent to frequency modulating the subcarrier with the flutter variations. By recording an unmodulated sub-carrier on a separate track on the tape recorder, a flutter only signal may be recovered upon playback for use in correcting the flutter corrupted data subcarrier.

A detailed block diagram of the correction scheme is shown in Figure 3-4. Further design details may be found in ref. 3, but have been omitted here as they do not substantially contribute to the theory of operation of the compensation technique. The basic theory behind the technique is to translate the flutter only signal to baseband (demodulate the subcarrier) by means of a "reference" discriminator. This signal is then used to vary the pulse width in the data discriminator detectors. When the "NOR" output is subsequently integrated (in the low-pass output filter), the result is as if no flutter had been present.

Figure 3-5 indicates the "NOR" output waveforms for the uncompensated and compensated situations. Figure 3-5-a shows the uncompensated case as was also seen in Figure  $3-2$ . Notice that in the "OFF" state  $(-9v)$ , the duration per cycle is constant due to a pulse being present with nominal width.

<span id="page-28-0"></span>



In Figure 3-5-b is shown the output of the "NOR" after the detector pulse widths have been modified. For the case where the recorder is running faster than nominal, the pulse width is compressed while when the recorder is running slower than nominal, the pulse width is expanded. Note that although the frequency of the resultant waveform varies as a function of the flutter, the output of the low-pass filter (an integrator) will be zero indicating that the flutter effect has been removed. It should be emphasized that the present discussion is concerned with a flutter only situation. In the presence of joint flutter and data information, only the flutter components are (theoretically) affected; the data are preserved intact.

At this point it is logical to discuss the prominent features of rack alignment as they relate to overall performance. The salient features of interest here are, 1) the scaling between the reference discriminator output and the extent of detector pulse width modification, and 2) the phase (or time) synchronization between the reference discriminator output and the detector outputs in the data discriminator.

With regard to the reference discriminator output scaling, consider the compensated data channel detector output pulse width to be represented by

# $T_{nominal}$  + K $V_{flutter}(t)$

where  $T_{nominal}$  is (as has been previously discussed) constrained to be onefourth the sub-carrier period and  $V_{f1utter}(t)$  is the output voltage of the low-pass filter in the reference channel. The parameter "K" is then the scaling required to match this voltage to the correct percentage expansion and compression of the detector output pulses in the data channel to effectively remove the flutter. If "K" is zero, no compensation has taken place. On the other hand if "K" is less than the correct value, insufficient compensation will occur. Further, if "K" is greater than the correct value, the data channel will be overcompensated and the flutter components enhanced. This behavior can be viewed as linear up to a saturation point where the stretched pulses begin to overlap or the compressed pulses begin to vanish. (This saturation effect was not considered during the study as this situation is considered non-normal.)

Temporal synchronization is required as a result of two features in the ITOS SR data processing channel. First, the flutter and data signals are

modulated on different subcarrier frequencies in the multiplex scheme and thus have different transport times through the MUX/DEMUX electronics. Second, the flutter baseband signal is obtained after the low-pass filter and amplifier in the reference channel discriminator and used to modify the data channel discriminator at the detector circuits. Thus the transport delay(s) of the "NOR", low-pass filter, and output amplifier of the reference discriminator must be incorporated. The majority of the synchronization is achieved by delaying the data channel prior to demodulation. A secondary adjustment is available in the reference discriminator. However, it appears to be a phase adjustment only and thus does not possess the range of the main delay. In order to effect proper cancellation, the time delay is adjusted to synchronize the reference channel and the data channel. In the event that the two channels become phased **180**° with respect to each other, a flutter enhancement occurs instead of cancellation. The extent of misalignment to cause enhancement is not considered likely at the ITOS identified flutter frequencies but could be significant if higher flutter frequencies were encountered.

#### 4.0 MODEL DESCRIPTION

<span id="page-31-0"></span>The basic approach to the model is that of a time domain simulation rather than development of a numerical treatment of theoretical transfer functions. This is largely the result of the non-linear functions encountered in the limiter, Schmitt Trigger, detectors, and the "NOR" circuits. The only exceptions to this design philosophy are the input band-pass filters and the lowpass output filters, where the frequency domain transfer functions were analytically specified and Fourier-transform, multiplication, inverse transform techniques were employed to avoid time-consuming numerical convolutions. Figure 4-1 shows an overall block diagram of the model implementation. The various approaches to each of the functional blocks are described in the following paragraphs. Table 4-1 indicates the parameter options incorporated into the model.

### 4.1 Reference Channel

The reference channel subcarrier frequency  $(f_0)$  is set at 6.25 Hz in order to be compatible with ITOS-D at 4:1 playback. The deviation index was selected to be  $+$  15% about this. The input band-pass filter was modeled about these parameters. Initially a theoretical complex frequency plane description for the filter transfer function was attempted but was rejected due to design complexity, and an ideal (rectangular amplitude and linear phase response) filter substituted. This has the effect of producing better transmission characteristics within the band-pass but introduces possible distortion as the base-band information contained in the higher order sidebands of the modulated subcarrier is rejected. This is considered to be negligible for the tone modulation used in the analysis.

The input amplifier, limiter, and cascade amplifier parameters were selected to be compatible with the EMR discriminator described in ref. 3-1.

The Schmitt Trigger (zero crossing detector) was forced to trigger on a threshold of + 1 mv for upcrossings and -1 mv for downcrossings to account for quantization noise contained in the (model) digitized frequency modulated subcarrier. A distinct deviation from the EMR discriminator design was the substitution of +9 volt level for an "on" condition and a -9 volt level for an "off" condition in the pulse stretching detectors (normal levels are -10 volts for "on" and -16 volts for "off"). This deviation resulted from a requirement to change pulse height instead of width in the compensation scheme in order to achieve

<span id="page-32-0"></span>

#### TABLE 4-1. PARAMETER OPTIONS

- <span id="page-33-0"></span>1. Transfer function of band-pass filter(s)
- 2. Input gain of limiter(s)
- 3. Clipping level
- 4. Output gain of limiter(s)
- 5. Inherent delay of limiter
- 6. Threshold of Schmitt Trigger(s)
- 7. Inherent delay of Schmitt Trigger
- 8. Pulse width scaling of detector (data only)
- 9. Pulse width of detectors
- 10. Output amplitude of detectors for "ON" condition ("ON" is pulse)
- 11. Output amplitude of detectors for "OFF" condition ("OFF" is no pulse)
- 12. Inherent delay of detectors
- 13. Output amplitude of "NOR" for "ON" condition
- 14. Output amplitude of "NOR" for "OFF" condition
- 15. Inherent delay of "NOR"
- 16. Transition voltage of "NOR"
- 17. Transfer function of low-pass filter(s)
- 18. Gain of output amplifier(s)
- 19. Main delay (error)

<span id="page-34-0"></span>finer resolution (i.e., each pulse width represents a small finite number of words in the algorithm, thus a small amount of width compensation would be absorbed in time base quantization effects). The "NOR" circuit was modeled as a logical equivalent of the EMR circuitry with the specified levels of  $\pm$  9 volts. Notice that the utilization of a "NOR" has the effect of doubling the subcarrier frequency and phase inverting the output signal (see section 3.0).

The output low-pass filter was modeled in the complex frequency domain and transform-inverse transform techniques employed. The filter has a 6 db/ octave roll-off and serves to integrate the output of the "NOR" circuit and to attenuate the contribution of the doubled subcarrier frequency.

The gain of the output amplifier was determined empirically for a single tone (see Section 5.0) and the polarity chosen to compensate for the phase inversion of the "NOR" circuit. Insofar as the amplifier is linear over its operating region the use of a single tone to determine the gain is justified.

An additional "delay" was incorporated into the model at the output of the reference discriminator to advance the reference signal with respect to the data subcarrier due to the delays encountered in the reference channel input bandpass filter and lowpass output filter. This would normally be achieved in the main delay in the data channel; however, in order to preserve the latter delay as a main delay "error", it was introduced at this point in the model.

#### 4.2 Data Channel

The data discriminator is modeled essentially in the same manner as the reference discriminator with the following exceptions. First, the data channel subcarrier was shifted to 25 KHz instead of 22 KHz utilized in the hardware. This was necessitated by the requirement that the detector output pulse width be one-fourth the period of the subcarrier frequency and the requirement that this time span occupy an integer number of words in the computer algorithm. Since the percentage deviation remained at  $+$  15%, this does not reflect any significant restriction on the model capability. Second, since the base-band signals and subsequently modulated subcarriers are generated within the algorithm, they are initially in phase and the requirement for a main delay per se does not exist. The main delay actually incorporated into the model

<span id="page-35-0"></span>represents a delay "error" and Is more meaningful in terms of overall rack operation. Third, the low-pass output filter was modeled (again in the complex frequency domain) as a seven pole passive filter with a roll-off of 42 dB per octave. This closely approximates the 40 dB/octave EMR specification. Fourth, the gain of the output amplifier was taken to be unity for convenience. This does not represent any compromise on model performance insofar as the areas of interest are represented by relative rather than absolute results.

# 4.3 Analysis Technique

The technique used for analysis of the results obtained from the model consisted primarily of the examination of flutter residuals and the sporadic harmonic behavior due to modulation non-linearities in the spectral domain. A sample output is shown in Appendix C. Additional comments regarding the specific analyses conducted are contained in the following section.
#### 5.0 SENSITIVITIES DERIVED FROM MODEL

The model was exercised parametrically to determine sensitivity to significant parameters. While the capability exists to analyse each of the parameters listed in Table 4-1, this study concentrated on the main delay and reference discriminator output gain settings. The reasoning behind this choice is twofold: one, the other parameters are considered to provide second order effects and, two, these two parameters represent critical adjustments in the z-axis rack alignment. These two parameters were initially tested by observing the degree of cancellation of a single tone imposed on both the data and flutter channels. Subsequently, analyses were conducted with multiple tones and random noise backgrounds.

Figure 5-1 shows the flutter residual after cancellation of a 1.2 count\* tone at 196 Hz as a function of delay error in micro-seconds. A change in delay of  $12.5$   $\mu$  sec would produce a change in flutter residual of approximately 0.02 counts at this frequency, representing a change slightly in excess of 1.5%. Since 25  $\mu$  sec is the resolution of the delay line, it would appear that the ability to achieve the proper delay is available (i.e.,  $+$  12.5  $\mu$  sec) providing the required total delay is within the range of the delay line (1575  $\mu$  sec). Notice that since what is being affected in the compensation is a cancellation effect, as the tone frequency increases, the apparent total phase change due to  $12.5 \mu$  sec error also increases. At 800 Hz (approximately the highest flutter frequency identified on ITOS-D), 12.5  $\mu$  sec would produce a phase change equivalent to 50  $\mu$  sec at 200 Hz. From Figure 5-1, one can roughly anticipate a flutter residual of 0.08 counts or about 6% of peak. Thus based on pure tone cancellation, the delay line adjustment resolution is adequate for z-axis operation. It will be seen later that the non-linearities associated with the modulation technique make the establishment of the proper delay setting difficult in an operational environment.

<sup>\*</sup>The term "count" represents the output of an eight-bit digitizer at DDHS which encodes the dynamic range of the sensor into a range of 256 levels. Strict details of this encoding are omitted here. It is sufficient to remark that low count values represent high temperatures while high count values represent low temperatures. Two counts are roughly equivalent to one degree Kelvin at the warm end and this scale factor is fairly linear over the temperature range of interest.



Figure 5-2 indicates the sensitivity of flutter residual as a function of reference discriminator output gain. Notice that zero gain represents no compensation and the original tone amplitude of 1.2 counts is preserved. In the range of zero to -0.21 the data channel is undercompensated while in the range of  $-0.21$  to  $-0.6$  overcompensation occurs. The gain is negative in sign to account for the inversion which occurs in the "NOR" circuit as described in Section 3.0. In the actual hardware this is achieved in the output amplifier while in the model it is achieved mathematically as the sign on the value input for the gain parameter. It should be noted that the absolute value of gain was derived empirically in order to effect model performance and may not reflect the actual hardware gain of the output amplifier. The value is indicative of relative behavior. No attempt was made to examine the saturation effect described in Section 3.0. As in the case of the delay setting, it will be shown later that an optimum gain setting will be difficult to achieve in an operational situation.

Table 5-1 shows the effect of modulating the data and flutter channels with four tones nominally identified to have the same frequency and amplitude as IT0S-D flutter components, and subsequently observing rack operation as the main delay is perturbed from its nominal setting. Notice the tone at 782 Hz deteriorates back to the no compensation condition more rapidly than lower frequencies. This is caused by a given delay representing a larger phase error at the higher frequencies. As a result, the no-compensation and flutter enhancement situations are reached sooner at the higher frequency tones. Notice that this behavior is similarly observed for tones at 102 Hz and 417 Hz. Note that a contrary effect is observed for the tone at 156 Hz where a delay error actually improves the rack performance until an error of 100 us is reached. This is considered to be due to the fact that due to the significant difference in tone amplitude, the tone is spectrally distributed differently and that in effect the non-linearity of the modulation technique has produced a phenomenon wherein the rack would appear to be exceeding the theoretical single tone performance. It should be commented that this type of data interaction was found to be prevalent in all multiple tone analyses. It is this type of behavior which causes the actual hardware to be difficult to align operationally.

Table 5-2 indicates the effect of varying the reference discriminator gain on the rack performance with the four tones described in the previous paragraph. Notice that the phenomenon of data interdependence is much more



# **TABLE 5-1. TONES ON DATA AND FLUTTER CHANNELS - DELAY ERROR EFFECT ITOS-D CONFIGURED AT 4:1**



**Values for no-compensation represent flutter tone amplitude in counts. The other values represent flutter residual in counts.**

**TABLE 5-2. TONES ON DATA AND FLUTTER CHANNELS - GAIN ADJUSTMENT EFFECT**



**Values for no compensation represent flutter tone amplitude in counts. The other values represent flutter residual in counts.**

**\*The values of gain are model dependent and should be interpreted only in a relative sense.**

radical here as evidenced by the (underlined) gain at which each tone is most effectively cancelled. In this case it is virtually impossible to define an optimum setting. Note that the various values tend to cluster around the -0.21 value obtained for the single tone at 196 Hz shown in Figure 5-2.

In view of the fact that the delay and gain adjustments are set simultaneously in an operational situation, the difficulty in identifying an optimum setting for delay and gain is further emphasized. In the model case, the situation is somewhat simplified in that the frequency modulated signals are initially in time synchronization. The only differential delay up to the point of compensation is due to the input band-pass filters of the reference and data discriminators. Since these have been modeled as ideal filters with linear phase response, the transit delay may be calculated exactly (the delay is given by the slope of the phase response). Thus in the model, lack of knowledge about the delay setting does not influence the gain setting analysis. However, in the hardware situation, the alignment resorts to an iterative process. It has been observed in actual testing that the main delay setting is the better initial parameter to adjust in achieving the final alignment.

Table 5-3 indicates an analysis where no tone modulation was applied to either the flutter or data sub-carriers and noise alone imposed on the flutter sub-carrier only. The intent here was to demonstrate the fact that noise on the flutter channel would result in increased noise level on the data channel output via the flutter compensation technique. The notation is defined as follows:

2<br>No = the noise variance of the data channel output with no compensation  $\sigma_{\mathcal{C}}^2$  = the noise variance of the data channel output with compensation

 $\sigma_{\bf n}^2$  = the contribution of noise in the data channel output due to noise on the flutter channel.

The data channel output has an apparent noise output due to quantization effects in the digital model. The important parameter to view here is  $\sigma_{\bf n}^2$  as a function of the bandwidth of the low-pass output filter on the reference discriminator. As would be expected, as the bandwidth is reduced, the noise variance reduces nearly proportionately.

Bandwidth of low- pass output filter	$\sigma_{NC}$	$\sigma$ <sub>C</sub>	σ n	
$2$ $kHz$	0.012	0.097	0.085	
$1$ $kHz$	0.012	0.053	0.041	
500 Hz	0.012	0.032	0.020	
250 Hz	0.012	0.021	0.009	
$100$ Hz	0.012	0.017	0.005	

**Table** 5-3. Noise on **Flutter** Channel - White, Uniform Dist.,  $\sigma^2 = 0.11$ .

 $\sigma_{\text{NC}}^2$  **-**  $\sigma_{\text{CC}}^2$  (since the noise samples are uncorrelated).







 $\sigma_{\text{NC}}^2 = 1.119$ 

**Values represent "dB Improvement" as defined in Appendix C.**

Table 5-4 shows the effect of re-instituting the four tones on the data and flutter channels in the presence of a noise background. The values presented here represent "dB improvement" as described in Appendix C. The intent is to show the trade-off of low-pass output filter bandwidth reducing the total noise on the data channel versus flutter cancellation effectiveness. Notice that as the bandwidth is reduced the cancellation ability degrades beginning at the higher tone frequencies as would be expected. The -1.0 "dB improvement" at the 417 Hz tone for a 100 Hz filter is in reality a flutter enhancement and is likely due to the odd spectral distribution of the modulated carrier which gives rise to the data interaction described previously. The 2 KHz filter provides superior cancellation performance to the lower bandwidth filters in all cases. However a reduction to 1 KHz reduces the overall noise variance and still provides acceptable cancellation. This kind of optimization of bandwidth versus performance would be more emphatic if the flutter noise was not white noise but was concentrated about certain flutter harmonics. Figure 5-3 shows the spectral distribution for the first 100 lines  $(\sim2440$  Hz) of the data channel output for the first three cases in Table 5-4. The flutter tones are indicated by the small diamonds below each plot. Of particular interest here is the behavior of the lines above the filter cutoff. For example, in Figure 5-3-c, wave numbers 90 and 93 have been significantly enhanced even when they are well in excess of the range of compensation. This again points out the strong data interaction due to the spectral distribution of the modulated carrier.

Finally, the effect of changing the roll-off rate of the low-pass output filter was investigated. Figure 5-4 shows the spectrum of Figure 5-3-b as compared with a similar situation where the order of the filter was increased from a single pole (roll-off rate of 6 dB/octave) to a two pole configuration (roll-off rate of 12 dB/octave). As would be expected, the total noise variance of the data channel was reduced from 0.197 to 0.170. This can be seen visually by inspecting Figure 5-4 in the region above the cutoff frequency (1000 Hz). However, notice that the cancellation effectiveness has decreased for the higher roll-off rate (note the increase in amplitude at those wave numbers marked with a diamond below each figure) . This would lead one to the conclusion that significant information relative to effective cancellation is contained in wavenumbers greater than the cut-off frequency





and that the lack of this information due to a higher rate of attenuation above cut-off noticeably affects performance.



Figure 5-4. Effect of Changirg Filter Roll-off.



# 6.0 PRELIMINARY RECOMMENDATIONS FOR DIAGNOSTIC IMPLEMENTATION

The philosophy of diagnostic implementation on-line at the DDHS is constrained to be a minimum interference technique. On a routine basis it must be rapid, repetitive, and not impose on computer storage requirements. Also, during those ingests where abnormal conditions are suspected, it must be of sufficient detail to be useful in correcting any system fault which may be present. These operational and performance constraints are best met here with an approach to the diagnostic implementation which is adaptive in the degree of detail as a function of the data quality observed.

Figure 6—1 shows a flow diagram of a recommended diagnostic procedure which would accommodate the above philosophy. As a simple measure of performance, an RMS would be computed on the space view each frame during ingest. This value would then be compared with a preset threshold. The threshold would represent the maximum noise level for which average or above average data quality could be assured. In the event the RMS was lower than the threshold, the ingest would continue and the operator advised of a normal condition at the termination of the ingest. In the event the RMS is greater than this threshold, it would be compared to a second (higher) threshold which would represent the maximum RMS noise which would assure the minimum acceptable data quality. If the RMS was lower than this threshold, a 64 sample power spectrum would be computed on the space view. The spectrum would be compared to a stored reference mask to see which flutter harmonics had changed and the result output for analysis. The ingest would continue and the operator advised of marginal data quality at the termination of the ingest. If, on the other hand, the RMS exceeded the maximum acceptable noise level, more detailed spectral analysis would be pursued. It would be ascertained whether nighttime visible data could be obtained within the remaining data to be ingested. If so, then a 1024 sample power spectrum would be computed and output for analysis and the ingest terminated. If not, the ingest would be terminated. The operator would be advised of the status at the termination of the ingest. A decision to reingest would then be made on the status observed and available spectral data.

During the course of the study, the complete diagnostic philosophy was not incorporated for on-line evaluation during ingest. However, an initial diagnostic configuration was developed and provided to NESS personnel for



Figure 6-1. Preliminary Diagnostic Logic Flow Diagram.

 $\blacksquare$ 

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preliminary simulation testing on the CDC 6600 at Suitland. The key elements of this configuration are shown in Figure 6-2. It is the intent of this type of approach to structure the final diagnostic configuration dynamically in order to maintain compatibility with system changes at NESS. In particular, it was not clear at the time of the conception of this configuration whether night-time visible data would be available, thus it has been deleted from the structure in such a way as to be included later if available.

The concept of performing a 64 sample power spectrum was given special attention in deriving the configuration shown. An analysis was performed to indicate the improvement in spectral stability which could be achieved by averaging. A vector of 1024 random samples with a flat spectrum was generated and its spectrum computed. This spectrum was then condensed to an equivalent 64 sample spectrum by summing the energy over groups of 16 wavenumbers. This result indicated a very flat spectrum as would be expected. The original 1024 sample data vector was then segmented into 16 independent 64-sample data sets and a spectrum computed for each. The results indicated these small sample spectra to be statistically unstable (i.e., flatness was not achieved). A cumulative average was then performed for the small sample spectra to determine the minimum number of spectra which could be averaged in order to reach a stable statistic. The average value at each wavenumber was computed with the number of spectra incorporated varying from 2 to 16. An apparently stable statistic was achieved at a minimum of six individual spectra. This cumulative average was then incorporated into the diagnostic logic.

The configuration shown in Figure 6-2 thus represents the basic cornerstone of the logic to achieve the final goal of stable compensation. It can be expanded and revised in accordance with ground system and satellite changes.



Note: Need to be sure not to compute PSD on all frames (i.e., upper threshold has not been incorporated) in the case of bad data.

Figure 6-2. Initial Diagnostic Configuration.

# 7.0 RECOMMENDED STANDARD OPERATING PROCEDURES

In order to provide a base for achieving stable overall system operation, a portion of the study was directed at preparing (in cooperation with Wallops CDA personnel) a "standardized operating procedure" for periodic alignment of the z-axis rack hardware. This effort resulted in an over detailed procedure which proved not to be viable on a day-to-day operational basis and was subsequently not formally adopted. It does represent a fairly complete technique for alignment and is included as Appendix E.



# 8.0 RECOMMENDATIONS FOR FUTURE EFFORT

It is recommended that the effort described in this report be continued toward the overall goal of achieving stable data quality assurance in an operational sense. Additional studies directed toward this end would include analytical investigations to quantitatively predict the effect of system changes (i.e. for future satellite configurations) on data quality and feasibility investigations to expand the scope of the diagnostic logic implementation. Additionally, close co-ordination and support is recommended during the actual operational implementation of this diagnostic logic at NESS.



# 9.0 APPENDICES

### APPENDIX A

## MODEL IMPLEMENTATION

This appendix presents a detailed block diagram of the digital computer algorithm which comprises the model developed during the study. A description of the model inputs follows as Appendix B, a sample output description as Appendix C, and a sample listing as Appendix D.

The block diagram indicates program control with the exception of the dotted line showing data flow for the flutter channel output. The subroutine call names are shown in parentheses. Power spectra are available as output at intermediate points in the program and are so indicated on the block diagram.



\*Power Spectra for Demodulated Signal(s)



## APPENDIX B

## MODEL INPUT DESCRIPTION

This appendix presents the configuration of the data cards used with the model. A separate data deck must be supplied for each pass through the algorithm. The data deck must be complete within itself, and no data cards are common to multiple passes. With proper care this provides maximum program flexibility in terms of conducting analyses. For the first two passes the input bandpass filters are bypassed, and cards No. 3 and 4 as well as 22 and 23 must be removed. The parentheses following the card number indicate the calling routine. The cards are numbered sequentially but with gaps in the numbering system due to data deck modification during development. The deck was not renumbered in order to avoid confusion over the span of the study. The numbers to the left of the description indicate card column.

# Data Deck Configuration



Card Number 9 (TRIGGR) - Integer Format - Diagnostic

- 1 Enter 1 if it is desired to output the data after the flutter channel has been examined for positive going and negative going zero crossings. 2 Enter 1 if it is desired to output the data which consists of the outputs of the A and B detectors. 3 Enter 1 if it is desired to output the data after the "NOR" circuit has "NORED" the outputs of the two detectors. Card Number 10 (TRIGGR) - "F" Format 1-10 The triggering level of the Schmitt Trigger for increasing voltages. II- 20 The triggering level of the Schmitt Trigger for decreasing voltages. 21-30 The delay associated with the Schmitt Trigger circuit. Card Number 11 (TRIGGR) - "F" Format 1-10 Enter 0.0 II- 20 Pulse width of detector A in seconds. 21-30 The delay in seconds associated with the A detector. 31-40 The amplitude of the detector A pulse output. 41-50 The amplitude of the detector A output when no pulse is being output. Card Number 12 (TRIGGR) - "F" Format 1-10 Enter 0.0. II- 20 Pulse width of detector B in seconds. 21-30 The delay in seconds associated with the detector. 31-40 The amplitude of the detector B pulse output. 41-50 The amplitude of the detector B output when no pulse is being output. Card Number 13 (TRIGGR) - "F" Format I- 10 The amplitude of the "NOR" circuit output voltage when it is in the high state. II- 20 The amplitude of the "NOR" circuit output voltage when it is in the low state. 21-30 The delay in seconds associated with the "NOR" circuit. 31-40 The input level which differentiates the low level from the high level. Card Number 14 (FLTRW) - Integer Format - Diagnostic
- 
- 1 Enter 1 if flutter array is to be output as it appears upon entry to the filter subroutine.
- 2 Enter 1 if flutter array is to be output after it is rearranged (put in complex form) suitably for use by the Fourier transform subroutine.

3 Enter 1 if Fourier spectrum of input data is desired.

4 Enter 1 if the spectrum of the filtered data is desired.

- 5 Enter 1 if the inverse Fourier transform output is desired.
- 6 Enter 1 if the flutter array is to be output after the filtering operations have ended.

Card Number 15 (FLTRW) - Integer Format

- 1-3 The degree of the transfer polynomial numerator for the output flutter channel filter.
- 4-6 The degree of the transfer polynomial denominator for the output filter in the flutter channel.
- Card Number 16 (FLTRW) 5E16.8
- 1-80 The coefficients of the numerator polynomial of the output flutter channel filter, power increasing except for the constant term which is last. Additional cards may be used for high order polynomials.

Card Number 17 (FLTRW) - 5E16.8

1-80 The coefficients of the denominator polynomial of the output flutter channel filter, power increasing except for the constant term which is last. Additional cards may be used for high order polynomials.

Card Number 18 (AMPL) - Integer Format - Diagnostic

1 Enter 1 if the arrays are to be output after the flutter array is amplified.

Card Number 19 (AMPL) - "F" Format

1-10 Enter the gain of the amplification stage.

Card Number 20 (DELAY) - Integer Format - Diagnostic

1 Enter 1 if the arrays are to be output after the flutter array is delayed (advanced, see Card No. 21).

Card Number 21 (DELAY) - "F" Format

1-10 The time in seconds by which the flutter channel signal is advanced with respect to the signal (effectively a positive error in main delay setting).

Card Number 21A (DELAY) - Integer Format - Diagnostic

1 Enter 1 if the arrays are to be output after the signal array is delayed (advanced, see Card No. 21B) .

Card Number 21B (DELAY) - "F" Format

1-10 The time in seconds by which the signal channel is advanced with respect to the flutter (effectively a negative error in main delay setting).

### Data Channel

 $Card$  Number 22 (FLTRA) - Integer Format - Diagn



- 2 Enter 1 if it is desired to output the data which consists of the outputs of the A and B detectors.
- 3 Enter 1 if it is desired to output the data after the "NOR" circuit has "NORED" the outputs of the two detectors.

Card Number 29 (TRIGGR) - "F" Format

zero crossings.

1-10 The triggering level of the Schmitt Trigger for increasing voltages.

11-20 The triggering level of the Schmitt Trigger for decreasing voltages.

21-30 The delay associated with the Schmitt Trigger circuit.

Card Number 30 (TRIGGR) - "F" Format

1-10 The ratio of the detector A pulse width output to the magnitude of the wow and flutter channel, i.e., the "coefficient" determined by

> detector pulse width = "coefficient"  $\times$  flutter magnitude + nominal detector pulse width.

- II- 20 The nominal pulse width detector A (see expression above).
- 21-30 The delay in seconds associated with the A detector.
- 31-40 The amplitude of the detector A pulse output.
- 41-50 The amplitude of the detector A output when no pulse is being output.

Card Number 31 (TRIGGR) - "F" Format

- 1-10 Ratio of the detector B pulse width to the magnitude of the wow and flutter signal. See corresponding entry in card number 30.
- II- 20 The nominal pulse width of detector B. See corresponding entry in card number 30.
- 21-30 The delay in seconds associated with the B detector.
- 31-40 The amplitude of the detector B pulse output.
- 41-50 The amplitude of the detector B output when no pulse is being output.

Card Number 32 (TRIGGR) - "F" Format

- I- 10 The amplitude of the "NOR" circuit output voltage when it is in the high state.
- II- 20 The amplitude of the "NOR" circuit output voltage when it is in the low state.
- 21-30 The delay in seconds associated with the "NOR" circuit.
- 31-40 The input level which differentiates the low level from the high level.

Card Number 33 (FLTRS) - Integer Format - Diagnostic

- 1 Enter 1 if signal array is to be output as it appears upon entry to the output filter subroutine. Useful for debugging only.
- 2 Enter 1 if flutter array is to be output after it is rearranged (put in complex form) suitably for use by the Fourier transform subroutine.
- 3 Enter 1 if the spectrum of data is desired.
- 4 Enter 1 if the spectrum of the filtered data is desired.
- 5 Enter 1 if the Fourier transform output is desired.
- 6 Enter 1 if the flutter array is to be output after the filtering operations have ended.

Card Number 34 (FLTRS) - Integer Format

- 1-3 The degree of the transfer polynomial numerator for the output signal channel filter.
- 4-6 The degree of the transfer polynomial denominator for the output filter in the signal channel.

Card Number 35 (FLTRS) - 5E16.8

1-80 The coefficients of the numerator polynomial of the output signal channel filter, power increasing except for the constant term which is last. Additional cards may be used for high order polynomials.

Card Number 36 (FLTRS) - 5E16.8

1-80 The coefficients of the denominator polynomial of the output signal channel filter, power increasing except for the constant term which is last. Additional cards may be used for high order polynomials.

Card Number 37 (AMPL) - Integer Format - Diagnostic

1 Enter 1 if the arrays are to be output after the signal array is amplified.

Card Number 38 (AMPL) - "F" Format

1-10 Enter the gain of the amplification stage.



#### APPENDIX C

### STANDARD OUTPUT FORMAT

This appendix indicates the data output format from the model. Also available is a description of the input parameters (which has not been included here). As described in Appendix B, intermediate diagnostic output is also available as an option.

The basic parameter of interest in model performance is "dB Improvement" across the spectral band. This is defined as

> A (without compensation)  $20 \log_{10} \frac{\text{h}}{\text{A (with compensation)}}$

where  $A_n$  is the amplitude of the "n"th harmonic in a spectral analysis of the data channel output. The spectrum for no compensation is generated during the first pass through the algorithm and stored. The above computation is provided for each subsequent pass desired.

Figure C-1 indicates the output format. The first column represents the wavenumber with wavenumber 1 being the "DC" term. The spectrum is computed using the Fast-Fourier transform method and incorporates 4096 data points at a sampling rate of 10  $\mu$  sec which corresponds to a  $\Delta f$  of 24.4 Hz at the DDHS. The second column represents the frequency in hertz at the spacecraft during record (i.e., real time), while the third column represents the frequency in hertz as seen at the DDHS during 4:1 playback from the CDA. (Recall that there is a 20.833:1 recorder speed increase during satellite readout.) The fourth column represents the spectral amplitude in counts for the no-compensation case and the fifth column the spectral amplitude in counts for the compensated case. Note that the flutter tones referred to in Section 5.0 appear at wavenumbers 5, 8, 18, and 33. The sixth column represents the "dB Improvement" mentioned above. In order to provide a feeling for what the magnitude of these numbers means, a 20 "dB Improvement" would indicate that the flutter harmonic has been reduced to one-tenth its original amplitude. Correspondingly, a 40 "dB Improvement" would indicate a reduction of the flutter harmonic amplitude to 1% of its original amplitude. Negative values indicate flutter enhancement. It should be noted that for the case presented,



Figure C-1. Sample Model Output Format.

the reference discriminator output gain was slightly misaligned resulting in a small "DC" component to be introduced into the data channel during the compensation process. For this reason the -146.753 "dB Improvement" of wavenumber 1 should be ignored. Columns 7 through 12 represent a continuation of the data presented in 1 through 6. Since 4096 samples were used in the spectral analysis, 2048 harmonics were generated with a Nyquist frequency of 50,000 Hz. This is well beyond the data channel bandwidth and only frequencies out to approximately 3000 Hz (wavenumber 120) at the DDHS were output. This adequately covers the band of interest for analysis of the z-axis rack operation.


## APPENDIX D

## SAMPLE PROGRAM LISTING

This appendix includes a sample listing of the digital algorithm used during this study. The model was programmed in Fortran IV - G level for an IBM Systems 370/65 located at the Triangle Universities Computation Center in the Research Triangle Park, N. C. The system log and source module map have been retained in the listing for completeness.





TFP237I 136 ALLOCATED TO PGPl^













 $71\,$ 













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ft












































#### APPENDIX E

#### STANDARD OPERATING PROCEDURES

As mentioned in Section 6.0, a detailed standard operating procedure for z-axis rack alignment was developed for review by CDA personnel. This appendix documents that procedure.

# Abbreviations

SR 1 = Scanning Radiometer Number 1

SR 2 = Scanning Radiometer Number 2

```
IR = Infrared
```
- VIS = Visible
- $F\delta W =$  Flutter and Wow
- SRR = Scanning Radiometer Recorder

 $S/C = Spacecraft$ 

MUX = Multiplexer

DEMUX = Demultiplexer

RT = Real Time (CDA Tape Recorder set to 60 ips)

ST = Slow Time (CDA Tape Recorder set to 30 ips)

```
A7 Control
```
Panel = Z-Axis Compensation Rack main control panel A7J3  $A^{7}J^{4}$  = Connectors or test points 3, 4, ..., 16 on control panel A7

```
A7J16
```
DVM = Digital Voltmeter (dc)

Note: The nomenclature RT F&W Discriminator #1, ST F&W Discriminator #2, etc. conforms to the RCA schematic for Rack 48 (last revision dated 5 December 1970). Connector designations are also based on this schematic.

#### List of Test Equipment



Z-Axis Alignment Standard Operating Procedures - Perform the following Daily Checks:

- (1) F&W Discriminator Zero Balance
- (2) Data Discriminator and VCO Balance

Initial Set-Up For Delay Adjustment - (This initial set-up should only have to be done once. If pre-pass checks start to vary, this operation should be redone after notifying Suitland)

- 1. ST Mode Reference Data (Figure E-l)
	- a. Put test tape on recorder and set for ST/Ch A playback and 15 ips.
	- b. Connect H-P 302A Wave Analyzer to DD#1 and tune to f<sub>fL</sub> (check with audio osc. and frequency counter). Turn off all F&W discriminators.
	- c. Adjust all delay line controls for minimum delay and set control panel for ST #1 operation.
	- d. Playback about 3 min. of tape and record 302A reading. Rewind tape to its starting point.
	- e. Tune HP 302A to  $f_{\text{fH}}$  (check with audio osc. and frequency counter) and repeat step l.d.
	- f. Connect HP 302A to DD *it*2 and repeat steps l.b. through l.e. with control panel set to ST *it*2 operation.
	- g. Set tape recorder for ST/Ch B playback and 15 ips.
	- h. Repeat steps l.b. through l.f.
- 2. ST Mode Alignment (Figure E-l with F&W Discriminators turned on).
	- a. Put test tape on recorder and set for ST/Ch A playback and 15 ips.
	- b. Set control panel to ST *it* 1 operation and all F&W discriminators turned on.
	- c. Connect HP 302A tp DD #1 and tune to f<sub>fH</sub> (check).
	- d. Start tape recorder and adjust delay unit *it* 1 until a minimum reading is obtained on the 302A. Record reading.
	- e. Rewind test tape to its start point.
	- f. Adjust 302A to  $f_{fI}$  and check.
	- g. Start tape recorder and record 302A readings. Let tape play through to the end. Rewind tape.
	- h. Set system for ST/Ch B playback.
	- i. Repeat steps l.a. through l.f. but DO NOT ADJUST DELAY LINE (just record readings on 302A).



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Use to Set Frequency of Wave Analyzer

cy Counter  $HP-5245L$ 

# Figure E-1. Initial Set-up for Delay Adjustment.

- j. Set system for ST/Ch A playback and 15 ips.
- k. Set control panel to ST #2 operation and all F&W discriminators turned on.
- 1. Connect 302A to DD  $#2$  and tune to  $f_{\rm{fH}}$  (check).
- m. Start tape recorder and adjust delay unit #2 until a minimum reading is obtained on the 302A. Record reading.
- n. Repeat steps l.d. through l.g.
- o. Tune 302A to  $f_{fH}$  and check.
- p. Start tape recorder and record 302A reading. Rewind tape.
- q. Adjust 302A to  $f^{\text{f}}_{\text{fI}}$  and check.
- r. Start tape recorder and record 302A readings. Let tape play to the end and rewind tape.

## Daily Check for Delay Adjustment

- 1. For data playback, connect as follows (Figure E-2).
	- a. Connect HP 302A to DD #1 and tune to  $f^{\text{H}}_{\text{fH}}$  (check).
	- b. Set Control Panel to ST  $#1/Ch$  A operation.
- **c.** Place test tape on tape recorder and set recorder to 15 ips.

## 2. Start tape recorder

- a. Adjust Delay Unit #1 for minimum reading on 302A (start with smallest delay increment, i.e. 25 µs).
- b. Record reading.
- c. Tune 302A to  $f^{\text{f}}_{\text{fI}}$  (check) and record reading.
- d. Stop and rewind tape.
- 3. Set Control Panel to ST #1/Ch B operation.
	- a. Tune 302A to  $f^{\text{fH}}$  (check).
	- b. Start tape recorder and record 302A reading. Stop tape.
	- c. Tune 302A to  $f_{fI}$  (check).
	- d. Start tape recorder and record 302A reading.
	- e. Stop and rewind tape.
- 4. Set Control Panel to ST #2/Ch A operation.
	- a. Connect 302A to DD #2 and tune to  $f^{\rm H}_{\rm{FH}}$  (check).
	- b. Start tape and adjust Delay Unit #2 for minimum reading on 302A (start with smallest delay increment, i.e., 25 ys).
	- c. Record reading.
	- d. Tune 302A to f<sub>fL</sub> (check) and record reading.
	- e. Stop and rewind tape.

5. Set Control Panel to ST #2/Ch B operation,

a. Repeat steps 3.a through 3.e.

6. Inform DDHS of any changes in Delay Unit Adjustments.

# ST OPERATING MODE



Use to Set Frequency of Wave Analyzer



## OTHER DAILY CHECKS

1. F&W Discriminator Zero Balance (Figure E-3)



Figure E-3. F&W Discriminator Zero Balance Test Set-Up.

- a. Set Audio Oscillator to 1 v. (rms) output and 6.25 KHz. + 20 Hz.
- b. Connect DVM to ST F&W Disc. No. 1.
- c. Switch Control Panel to ST #1 Channel A operation.
- d. Adjust BAL control on front panel of Model 287T Channel Selector Plug-In of ST #1 F&W Reference Discriminator for 0 + 0.005 volts dc on DVM.
- e. Set BANDEDGE VOLTS Control on RT #1 F&W Reference Discriminator fully clockwise.
- f. Connect DVM to ST F&W Disc. No. 2.
- g. Switch Control Panel to ST #2.
- h. Repeat 1.d. and 1.e. for ST #2 F&W Reference Discriminator.

2. Data Discriminator and VCO Balance (Figure E-4)



Figure E-4. Data Discriminator and VCO Balance Test Set-Up.

- a. Turn ST #1 and ST #2 F&W Discriminators off. Use DVM on their outputs (see 1. for connectors) to verify 0 volts dc output.
- b. Set Audio Oscillator to 1 volt (rms) output and 22.8 KHZ. + 20 Hz.
	- (1) Switch Control Panel to ST #1.
	- (2) Connect DVM to DD #1.
	- (3) Connect second H-P 5245L Counter to output of VCO #1.
	- (4) Adjust BAL control of ST #1 Data Discriminator for 0 + .005 volts dc on DVM.
	- (5) Adjust FREQ. Control of VCO #1 for a frequency output of 22.8 KHz. + 20 Hz. on second counter.
	- (6) Set Audio Oscillator to 1 volt (rms) output and 26.4 KHz + 20 Hz. (UBE)
	- (7) Adjust BANDEDGE VOLTS control on ST //I Data Discriminator for +5.0 volts dc on the DVM.
	- (8) Adjust INPUT control of VCO #1 for an output frequency of  $26.4$  KHz.  $+$  20 Hz. (UBE) on second counter.
	- (9) Set Audio Oscillator to 1 volt (rms) output and 19.2 KHz. + 20 Hz. (LBE)
- (10) Reading on DVM should be approximately -5.0 volts (record reading).
- (11) Record frequency out of VCO #1 (should be 19.2 KHz.  $+$  20 Hz. LBE).

c. Set Audio Oscillator to 1 volt (rms) output and 22.8 KHz.  $+$  20 Hz.

- (1) Switch Control Panel to ST #2.
- (2) Connect DVM to connector DD *it*2.
- (3) Connect second H-P 5245L counter to output of VCO *it*2.
- (4) Repeat 2.b.(4) through 2.b.(ll) on ST #2 Data Discriminator and VCO *it2.*

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## Master Test Tape Recording

1. In order to align the z-axis rack, it is required that only a single tone be recorded on the visible and/or IR tracks of the SRR. This means that the portion of the SR Processor which multiplexes the SR output with telemetry, step wedge, etc., data must be defeated. 2. The two equipments which will lead to the greatest time delay difference between the SR IR/Visible channel and the F&W channel are the spacecraft multiplexer and the ground-station demultiplexer. In recording the master tape, it is required that MUX, DEMUX and 14 track tape recorder be as similar as possible to the actual operational equipment.

3. The diagram in Figure E-5 indicates roughly how the tape is to be made. No Visible, Data or telemetry are shown since none of these signals are desired. No particular attention has been given to levels; however, attenuators or amplifiers may be necessary. The four cables from the output of the DEMUX to the 14-track tape recorder should all be of equal length. If possible, a virgin tape should be used on the 14-track recorder and it should be completely filled (endto-end on tracks 5, 7, 9, 11) with test tone data. The two most important points to remember about this test tape are

(a) Continuous tones must be recorded on tracks 5, 7, 9, 11 of the 14-track recorder. Thermal vacuum data are not sufficient since they are not constant in amplitude due to the time-multiplexed data on the SR video output.

(b) Short of changing the spacecraft connections to eliminate unwanted data, the system should be as close as possible to operational conditions.



Master Tape Recording Set-Up. Figure E-5. Master Tape Recording Set-Up.Figure E-5.

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## 10.0 REFERENCES

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- 2-2. Black, H. S.: Modulation Theory. Van Nostrand, 1962.
- 3-1. EMR Model 287A-02 Subcarrier Discriminator (Revision E) Instruction Manual, EMR Telemetry, April 1970.