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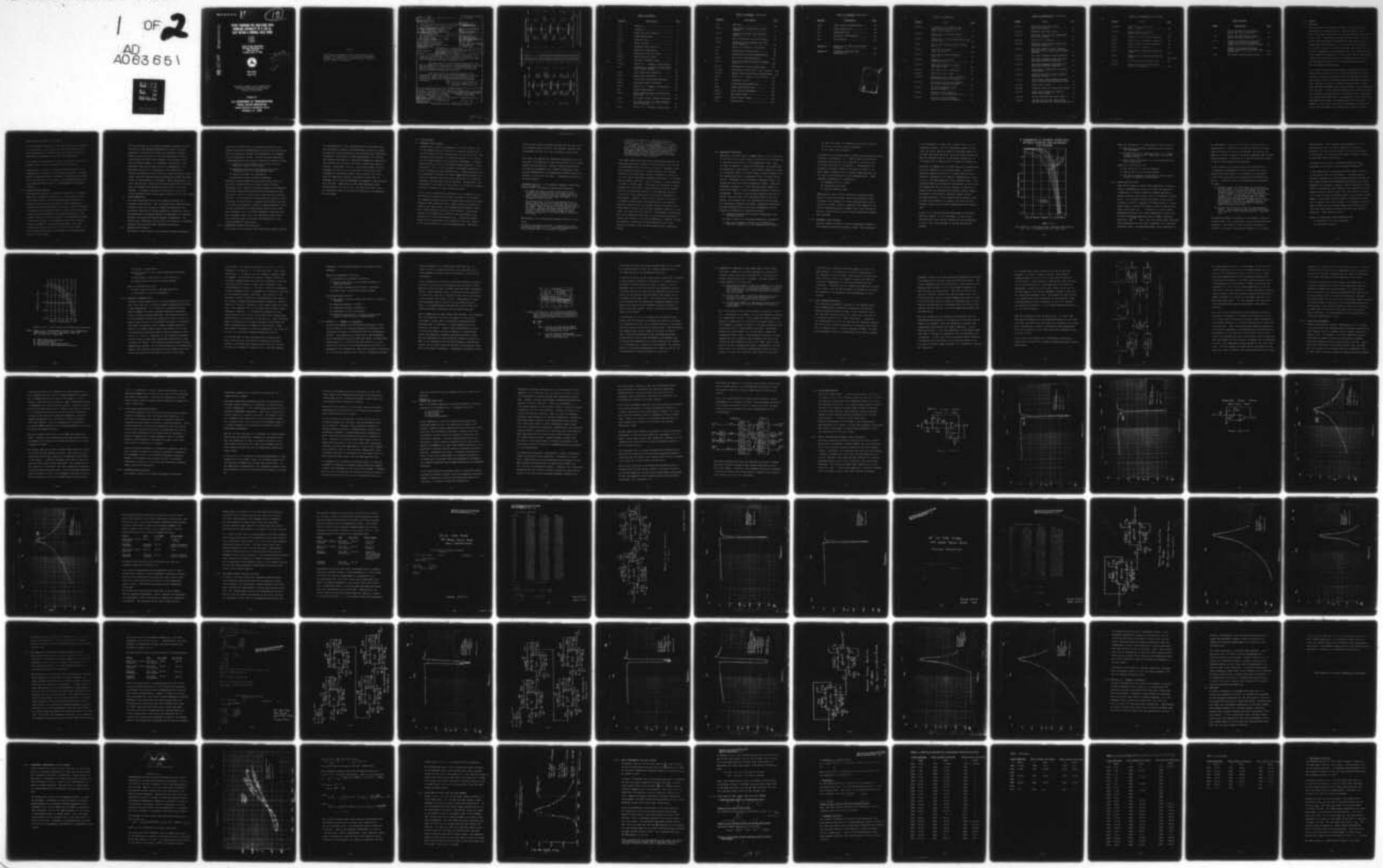

Report No. FAA-RD-78-116

STUDY PROGRAM FOR ANALYZING DATA HANDLING CAPABILITY OF A 240 Hz. SLOT WITHIN A NORMAL VOICE BAND

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16. Abstract

The program covered a research phase as well as a laboratory testing phase, to determine the feasibility of handling data in a 240 Hz slot within the normal voice band over an unconditioned 3002 transmission line.

The goals were to determine the maximum data rate and the effects on voice quality. The maximum data error rate was specified as 1×10^{-4} (.0001), which included choosing a mod./demod. technique, and the design of filters for notching the voice band and bandpassing the data.

Both theoretical and practical aspects of implementing a data slot technique are discussed, along with schematics, tables and graphs illustrating filter, modulation and theoretical characteristics of the applied technique.

(6) Study Program for Analyzing Data Handling Capability of a 240 Hz. Slot within a Normal Voice Band.

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METRIC CONVERSION FACTORS

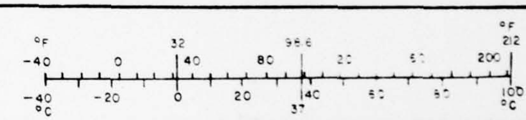
Approximate Conversions to Metric Measures

Symbol	When You Know	Multiply by	To Find	Symbol
		LENGTH		
in	inches	*2.5	centimeters	cm
ft	feet	30	centimeters	cm
yd	yards	0.9	meters	m
mi	miles	1.6	kilometers	km
		AREA		
in^2	square inches	6.5	square centimeters	cm^2
ft^2	square feet	0.09	square meters	m^2
yd^2	square yards	0.8	square meters	m^2
mi^2	square miles	2.6	square kilometers	km^2
	acres	0.4	hectares	ha
		MASS (weight)		
oz	ounces	28	grams	g
lb	pounds	0.45	kilograms	kg
	short tons (2000 lb)	0.9	tonnes	t
		VOLUME		
tsp	teaspoons	5	milliliters	ml
Tbsp	tablespoons	15	milliliters	ml
fl oz	fluid ounces	30	milliliters	ml
c	cups	0.24	liters	l
pt	pints	0.47	liters	l
qt	quarts	0.95	liters	l
gal	gallons	3.8	liters	l
ft^3	cubic feet	0.03	cubic meters	m^3
yd^3	cubic yards	0.76	cubic meters	m^3
		TEMPERATURE (exact)		
°F	Fahrenheit temperature	5/9 (after subtracting 32)	Celsius temperature	°C

*1 in = 2.54 (exactly). For other exact conversions and more detailed tables, see NBS Misc. Publ. 286, Units of Weights and Measures, Price $2.25, SD Catalog No. C13.10:286.

Approximate Conversions from Metric Measures

Symbol	When You Know	Multiply by	To Find	Symbol
		LENGTH		
mm	millimeters	0.04	inches	in
cm	centimeters	0.4	inches	in
m	meters	3.3	feet	ft
m	meters	1.1	yards	yd
km	kilometers	0.6	miles	mi
		AREA		
cm^2	square centimeters	0.16	square inches	in^2
m^2	square meters	1.2	square yards	yd^2
km^2	square kilometers	0.4	square miles	mi^2
ha	hectares (10,000 m^2)	2.5	acres	
		MASS (weight)		
g	grams	0.035	ounces	oz
kg	kilograms	2.2	pounds	lb
t	tonnes (1000 kg)	1.1	short tons	
		VOLUME		
ml	milliliters	0.03	fluid ounces	fl oz
l	liters	2.1	pints	pt
l	liters	1.06	quarts	qt
l	liters	0.26	gallons	gal
m^3	cubic meters	35	cubic feet	ft^3
m^3	cubic meters	1.3	cubic yards	yd^3
		TEMPERATURE (exact)		
°C	Celsius temperature	9/5 (then add 32)	Fahrenheit temperature	°F



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1.0 GENERAL

1.1 OBJECTIVE

The objective of the 240 Hz Slot Study is to determine the feasibility of handling data in a 240 Hz frequency slot within the normal voice band over an unconditioned 3002 transmission line. The program covered the notch effect on voice quality and determined the maximum data rate that could be transmitted and received with a bit error rate efficiency of 1×10^{-4}, and with a push-to-talk send and confirm time of 80 msec, independent of transmission link. The study covered both a theoretical approach, as well as the use of experimental circuits for laboratory testing, to determine the data rate and the system constraints of an integrated voice channel/ data sub-channel configuration.

In the basic slot signalling approach, band reject (notch) filters are placed in the voice path to eliminate voice frequencies in the band assigned to data carrier and to remove data carrier from the audio delivered to the receive terminal. Also, bandpass filters are placed in the data path to band limit the data frequency spectrum and separate voice components from the data. In the laboratory model, voice and data signals were combined, transmitted over an artifical line (simulating and unconditioned 3002 line), and separated at the receive end. Measurements were made to determine data rate capability. In addition, voice tapes were made to subjectively evaluate the effect the attenuated voice frequencies, within the notch, would have on overall voice quality.

By utilizing this theoretical and laboratory testing approach, it was possible to investigate not only the feasibility of data transmission in a 240 Hz slot, but also the practical implementation aspects of this type of transmission.

The results of the laboratory tests, along with the test procedures, can be found in appendix A of this report.

Additionally, in order to provide the maximum amount of information as it pertains to the feasibility study, a minimum amount of text book and/or documented literature type information is contained in this report. For information relating to the details of the theory, a bibliography is provided in Section 6.

1.1.1 SCOPE AND STUDY SUMMARY

The scope and study of the program included both a research phase and a laboratory testing phase. The research phase covered the study of the various modulation techniques to determine the most efficient means of data transmission, considering the slot bandwidth limitations, and slot location considerations for minimum effect on voice quality and interference to Single Frequency (SF) signalling signals. The primary source for research material was obtained from Drexel University's Library in Philadelphia, while other research sources included Dr. Peter Hahn (Consultant), Mr. Bruce Fritchman of Lehigh University and Orion's own library. A list of research documents is provided in the bibliography section of this report.

The testing phase of the program included the design and construction of PSK modulator/demodulator circuits, filters, amplifiers, line drivers, buffers and interface circuits. These circuits were used to test the feasibility of transmitting data and voice simultaneously, with selected voice notch and data bandpass in a laboratory environment. Consequently, the circuits were built in a breadboard or experimental form for use primarily for this application. Further development of these circuits, which is beyond project scope, would be required before any practical system implementation or testing could be considered. These experimental circuits did, however, provide the necessary means for collecting data to determine the feasibility goals of the slot study program. A schematic diagram of the circuits, as well as filter characteristics, are provided in Section 2.5 of this report.

1.2 STUDY METHODOLOGY

The study methodology used on this program consisted of a two-phase investigation. The first phase was a research phase (analysis task) and directed at conducting quantative and qualitative analysis of system constraints, parametric limits, and operational techniques through the employment of literature research and analytical/system synthesis methodology. The second phase (laboratory testing task) was directed at recording empirical data measured under laboratory conditions.

1.2.1 ANALYSIS TASK (Phase I)

The analysis task portion of the program covered approximately six weeks of effort,with the research directed not only toward theoretical implementation of data in the 240 Hz slot, but also toward the pragmatic implementation of equipment into an operational system. The key technical parameters investigated during this period covered the following areas:

(1) Location of the frequency slot within the bandwidth of a normal voice channel.

(2) Modulation techniques and selection of one for incorporation into an experimental model.

(3) Analysis of data rate performance.

(4) Analysis of speech quality performance.

Analysis of these parameters,along with their effects on practical considerations for devices such as filters, amplifiers, etc., required extensive research of documented literature (Drexel University and Orion's Libraries) with relation to the slot constraints in both the voice and data channels. It was difficult, especially with the short time constraint of the contract, to obtain data directly related to the 240 Hz requirement; however, numerous documents covering data transmission techniques over the full voice band were available, and it was from information extracted from these documents which led to the selection of the voice/ data technique used to determine the feasibility of the limited bandwidth transmission.

1.2.2 LABORATORY TESTING TASK (Phase II)

The results of the analysis task were used as design criteria for implementation in the second phase of the program which was directed at designing and developing circuits that could be used for recording empirical data measured under laboratory conditions. This effort covered approximately 4-1/2 months and included the design of a modulator/demodulator, amplifiers and line drivers, and both band reject (notch) and bandpass filters. Using these circuits, a feasibility model was built and empirical data was collected which included both voice recording of transmitted speech, with a notch in the voice band for subjective analysis, and data bit error rate recordings, with a limited bandwidth, of random bit data patterns up to 63 bits. Additionally, tests were performed using fixed patterns of 4 and 8 bits. The results of these tests are included as part of the Test Plan covered in the Appendix of this report.

2.0 STUDY ANALYSIS

2.1 FREQUENCY SLOT LOCATION

As previously stated in paragraph 1.2.1 of this report, one of the primary areas of concern that required investigation was the location of the frequency slot in the voice band. The frequency slot allocation is important because of its effect on both speech quality and operational performance. Although speech quality is not a well defined speech parameter - hence subjective - it is a real parameter and probably important in speaker recognition and intelligibility. These characteristics are important in aircraft and tower communications because variations in speaker voices may be an indication of a potential emergency, and therefore must be detected. Consequently, the requirements of the FAA are that little or no effect be imposed on either of these two speech characteristics.

In the area of operational performance, it is required by the FAA that the transmitted data have no effect on or interfere with existing in-band signalling systems which utilize a Single Frequency (SF) signalling tone. These tones are primarily at 2600 Hz, but some 2-wire trunk systems may use 2400 Hz; therefore, these frequency areas are to be avoided. And finally, because of the frequency roll-off at the high frequency extremities of a 3002 unconditioned line, a potential problem exists in the transmission and reception of the data signals if the slot is placed at the high frequency end. The FAA is already experiencing equipment problems at the high end of the frequency band because of the high end roll-off, and recommended that the slot not be located in this range.

With these two operational constraints imposed on the slot location, it became obvious that the slot had to be located between 300 and 2350 Hz. However, further consideration provided by recommendations of Comité Consultatif International Télégraphique et Téléphonique (CCITT) limited the placement of the slot even more. Consideration was given to the following:

Recommendation Q.22 - Fifth plenary assembly (Green Book)
Vol. VI-1 page 59
Titled: CCITT - Telephone Switching and Signalling

> "To reduce the risk of signal imitation by speech currents the frequencies for an in-band signalling system should be chosen from the frequencies in the band in which speech signal power is lowest, i.e., frequencies above 1500 Hz.
>
> The desirability of this was confirmed by tests in London, Paris and Zurich in 1946 and 1948 to choose the signalling frequencies of systems standardized by the CCITT. These tests led to the conclusion that if relative freedom from false signals was to be obtained, other than by undue increase in signal strength, frequencies of at least 2000 Hz would have to be used."

Additionally, the following Bell System criteria is also cited:

Preliminary Technical Reference - "Acoustical Coupling for Data Transmission" - Bell System Data Communications, dated November 1968, which states as follows:

"--in order to prevent the interruption or disconnection of a call, or interference with network control signalling, it is necessary that the signal ---- coupled to the Telephone Company ---- should at no time have energy in the 2450 to 2750 Hz band ---- power is in the 2450 to 2750 Hz band, the resultant electrical power generated at the telephone set in the 2450 to 2750 Hz band should be less than the corresponding electrical power present at the same time in the 800 to 2450 Hz band."

With these operational performance constraints defined, the remaining constraint (effect on speech quality) will now be the determing factor for the slot location. Research information obtained during the research phase of the program, indicates that most speech energy information is located at the lower end of the speech frequency spectrum, i.e., between 300 and 1500 Hz (22)(24). This, of course, is also confirmed by CCITT as stated above. Unfortunately, the elusive speech parameter(s) which make up "speech quality" are not as well defined, and reference material in the area was, at best, difficult to find, especially with the time constraint of the program. However, since the slot location is limited to a frequency range between 2000 and 2350 Hz, and reference material indicates that most speech energy is located in the lower frequency band, then the slot placement should be in the upper voice band but below 2350 Hz. Consequently, considering the 240 Hz bandwidth limitation, a notch center frequency of 2190 Hz was chosen for the data transmission. However, because of component tolerances, the data modulator circuit was actually centered at 2210 Hz when measured with a frequency counter.

2.2 MODULATION TECHNIQUES

Typically, as in any digital communication, or as in the case of the 240 Hz slot study, rarely is the data to be transmitted in the form best suited for the transmission link. Hence, in order to enhance transmission efficiency, the signals must be translated in some manner. This process, whereby a signal is transformed from its original form to a more suitable form for transmission over a transmission link, is called "modulation." Depending on the modulation technique chosen, the process may shift the signal frequency or phase, change the bandwidth occupancy, or it may even materially change the form of the signal in order to optimize noise or distortion performance. However, no matter what modulation process is chosen, there are only three approaches used to vary the carrier in accordance with the information to be conveyed; viz, amplitude, frequency and phase. Modulation techniques are then developed using these basic approaches with the objective of meeting the following criteria:

(1) Generate waveforms which differ recognizably from prevailing noise.

(2) That a high rate of sending information is possible.

(3) That the sequences of digits are represented as waveforms acceptable to the communications channel.

(4) That the signal is suitable for non-ideal channels.

(5) Exploit available channel bandwidth.

(6) Be simple and inexpensive.

To examine the total multitude of modulation techniques and combination of techniques, e.g., combined amplitude and phase modulation and how they can be applied to the slot study, was beyond the scope of the project. Consequently, for purposes of brevity, emphasis was placed on three of the more common techniques used in digital communications, and one that is relatively new in data communications. These candidate techniques are respectively:

(1) Frequency Shift Key (FSK)

(2) Phase Shift Key (PSK)

(3) Vestigal Sideband (VSB)

Additionally, each of these modulation techniques lends itself to multilevel schemes which would allow the basic technique to be expanded if future considerations were required by the FAA. However, because of the complexity and cost of implementing these schemes, and in the interest of brevity, only the basic forms of these candidate techniques were examined.

2.2.1 FREQUENCY SHIFT KEY(FSK)

The most popular technique in data transmission systems is the frequency modulation approach or FSK. The information to be transmitted is coded into a binary form using two elementary signals. The two elementary signals, called "one" and "zero" (mark and space in telegraph terms) are of equal and finite duration, occur with equal probability and are differentiated by two different frequencies, or if multiple FSK is employed by "n" different frequencies.

Detection or demodulation of these signals is accomplished by either coherent or non-coherent means. A coherent detection system is defined as a system which utilizes a reference frequency at its receiver (demodulation), which is set to the same frequency of the incoming signal. This technique makes the system less susceptable to noise since the demodulator does not make a decision to provide an output unless the incoming and reference signals compare. Conversely, a non-coherent detection system utilizes envelope detection, and the decision as to whether a mark or space was transmitted is made on the basis of which detector output has the highest amplitude at the sampling time.

Figure 2.2.1-1 illustrates the advantages of a coherent detection system. As is illustrated, it can be seen, theoretically, that coherent systems are less susceptable to noise, even when extended to include multiple FSK systems.

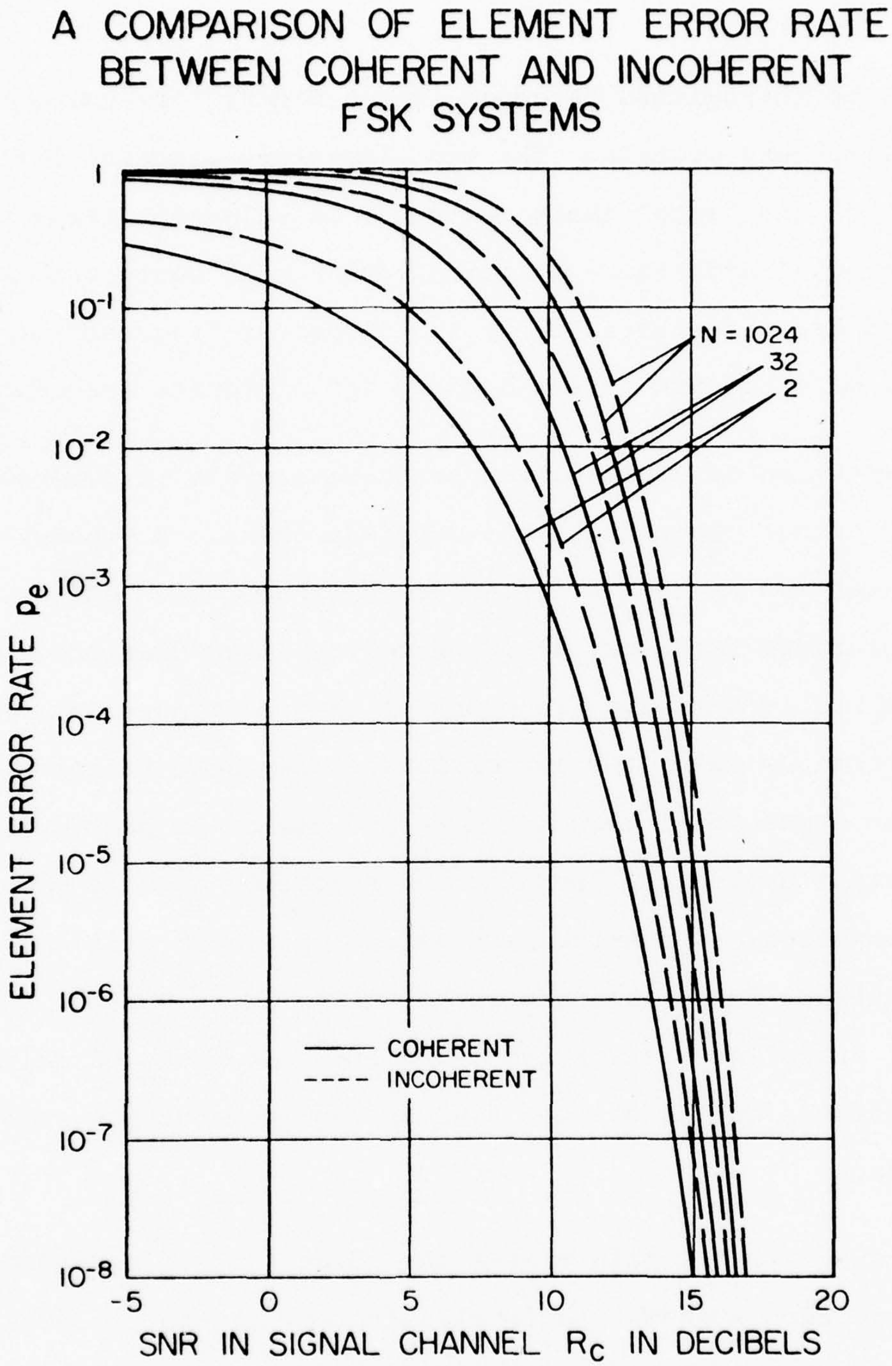


Figure 2.2.1-1

From: Akima, H., "The Error Rates in Multiple FSK Systems," Nat'l Bur. of Standards Note 167, March 1963.

Among the advantages of an FSK system are the following:

(1) Because of its history, much data can be found on its performance.

(2) Because each bit is relatively long, it is immune to many types of errors, especially with coherent detection schemes.

(3) With today's technology, it is cost effective and easy to implement.

Among its disadvantages are:

(1) Not efficient in terms of bandwidth.

(2) Used primarily in low speed systems.

(3) Not easily adaptable to multilevel coding schemes because of bandwidth inefficiencies.

2.2.2 PHASE SHIFT KEY (PSK)

Phase shift keying or digital phase modulation, transmits digital information by using, as a code, the sequential transmission of carrier pulses of constant amplitude, angular frequency and duration, but of a different relative phase. In a two-phase system, one phase of the carrier frequency is used to represent one binary state (1), while another phase, usually 180^{o} apart is used to represent the other binary state (0). In addition, it can be shown mathematically that binary phase modulation is equivalent to Double Sideband Suppressed Carrier (DSBSC) amplitude modulation (AM)[25]. Thus, since the digital phase modulated signal possesses the same basic properties as an amplitude modulated signal, the modulated signal can be generated in the same manner. This provides a cost effective means of implementation. Furthermore, it has the advantage of bandwidth utilization as found in AM systems, as well as high immunity to noise as found in frequency modulated (FM) systems, thereby providing a combination of attractive features found in each of these modulation techniques.

In the recovery or receiving section of a PSK system, it is necessary to demodulate the carrier signal by using a coherent detection scheme if the system is to capitalize on the error rate efficiently provided by this modulation method. A coherent detection technique can be provided in two ways:

(1) Provide a phase reference signal at the receiver. The two phases, as in a binary system, are detected by multiplying the information signal with this reference signal, which is of the same frequency as the incoming carrier and with a known phase with respect to it. The reference signal is usually arranged to be exactly in phase with one of the binary signals and 180° out of phase with the other. This method is difficult to implement because of synchronization, but does provide an efficient means of data recovery, and

(2) Extract the reference carrier from the modulated signal. Some techniques used to extract the signal are: squaring, Costas loop and decision - directed loop.

The squaring method as employed on this program, feeds the incoming signal into a multiplier circuit which doubles the frequency. This signal is then filtered and applied to divide by 2 circuits, returning the signal to its original frequency rate. This signal is then compared to the incoming signal, which has been clocked by the doubled frequency, and the resultant signal is the recovered data. In this manner, the input carrier signal is used as a means of providing coherent detection.

Information on the Costas and decision-directed loop techniques can be found in the existing literature[26].

In some applications, obtaining a coherent phase reference at the receiver may not be feasible. In such systems, a practical variation of coherent detection may be implemented. This method called "phase-comparison detection," uses phase comparison of successive samples. With this process, information is provided by phase changes of the transmitted signal between adjacent signal pulses; thus, the receiver circuits do not need to know the absolute phase of the incoming signal. However, because the reference phase is disturbed by noise just as that of the carrier signal, it is apparent that this type of detection will produce higher error rates for any specified input signal-to-noise ratio than with coherent detection. (Ref. Figure 2.2.2-1).

Among the advantages of PSK modulation are:

(1) Insensitive to level variations.

(2) High noise immunity.

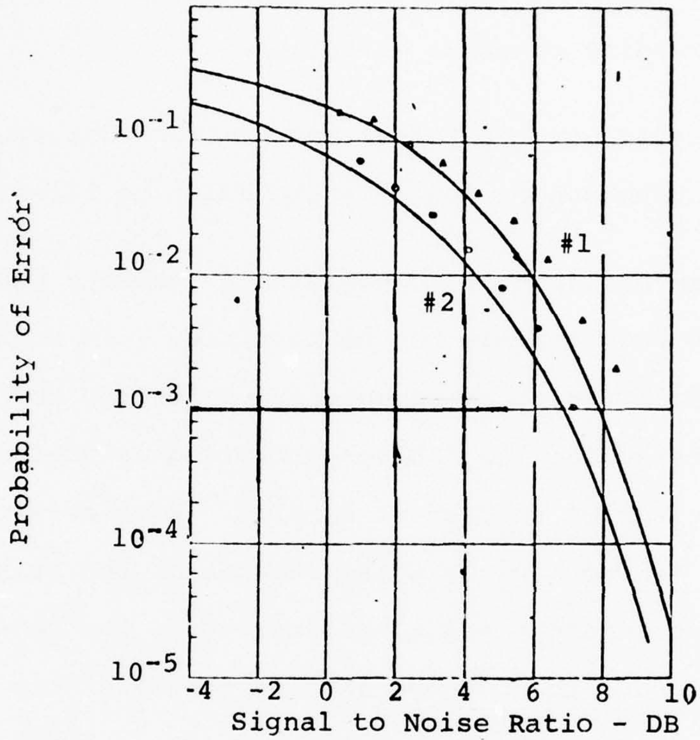


Figure 2.2.2-1 - Error rates with binary phase modulation.

From: Cahn, C. R., "Performance of Digital Phase-Modulation Communications Systems," IRE Trans. Comm. Sys. Vol. 1, CS-7, No. 1, May 1959

#1 Phase Comparison Detection
#2 Coherent Detection
ο Experimental Coherent Detection
▲ Experimental Phase-Comparison Detection

(3) History of performance.

(4) For a given bit rate, better bandwidth utilization than FM.

(5) With today's technology it is cost effective.

(6) Lends itself to multilevel coding schemes.

Among its disadvantages are:

(1) More complex than FSK or AM type modulation.

(2) Susceptable to phase disturbances.

2.2.3 VESTIGIAL SIDEBAND (VSB)

Vestigal Sideband Modulation is a form of amplitude modulation and is the result of an effort to improve bandwidth efficiency of AM transmission. It is basically a compromise between double sideband and single sideband modulation techniques. Since in double sideband modulation, each sideband contains all the essential information required for data recovery, it appears obvious that if one sideband is removed the bandwidth efficiency is improved; hence, single sideband modulation. However, in digital communication, the carrier is lost with this method when continuous mark or space transmission takes place; i.e., the d.c. component of the information envelope is lost, since no modulation takes place except when a pulse is started or ended. This condition results because in data communication systems the on and off pulses are the modulating signals, and modulation products which are formed occur only during the transition time from on to off or vice versa.

Consequently, the modulation products are transients whose bandwidth is a function of the switching rate. Under these conditions, it is apparent that the complete sideband cannot be removed. To overcome this problem, digital systems use vestigial sideband modulation where one sideband is retained, a portion of the carrier is kept, and a "vestige" of the other sideband is retained. In order to form this compromise, a double sideband signal is produced and filtering is performed on the unwanted sideband components. The results of this operation is that a bandwidth of approximately three-fourths of a double sideband modulation system is acquired. However, because of the lack of symmetry of the signal caused by filtering, a wave component 90° out of phase with the basic signal - quadrature component - is produced and combined with the signal which in turn distorts the digital pulses making them more difficult to reconstruct at the receiver. This difficulty adds to vulnerability of changes inherent in AM systems and makes the system even less immune to interference. SAGE data systems confirmed, by extensive experience, the sensitivit_ to noise of a vestigial sideband transmission system.

To overcome some of these problems found in VSB systems, pilot tones are often transmitted to maintain continuous control of the phase and frequency error, and d.c. restorer circuits are included to restore the d.c. and low frequency components of the passband signal to facilitate carrier recovery.

Among the advantages of VSB are:

(1) Efficiency in bandwidth utilization.

(2) Require less correction for delay distortion in high speed system.

(3) Relatively uncomplicated and easy to maintain.

(4) Lends itself to multilevel coding schemes.

Its disadvantages include:

(1) May require auxiliary pilot tones and d.c. restorer circuits.

(2) Highly susceptable to noise.

(3) Quadrature component distortion.

(4) Vulnerable to level variations.

(5) History of performance - relatively little data could be found during the research phase.

2.2.4 SECTION 2.2 - SUMMARY & CONCLUSIONS

As can be determined from the previous paragraph in this section, each of the candidate modulation techniques have "something to offer" in conjunction with the transmission of data. The problem was to chose one of the candidates which would provide the most efficient means of transmission (design goal of 1×10^{-4} for bit error rate), at a rate which would allow for 80 millisecond send and confirm, independent of the transmission link, with the major constraints of (1) a 240 Hz frequency slot, and (2) a transmission media which consisted of an unconditioned 3002 wireline. In order to meet the program objective, the pros and cons of each candidate technique had to be assessed in terms of the constraints.

First consideration was given to the FSK technique, primarily because of its history of performance and its ease of implementation with the advent of Phase Lock Loop (PLL) technology. However, when compared to PSK, it did not utilize bandwidth as efficiently, nor did it perform as well for a given signal to noise ratio, Ref. Figure 2.2.4-1. Additionally, present day technology did not justify the cost effectiveness of PLL's since PSK generators and detector circuits were also readily available, therefore, PSK was chosen over FSK.

When a comparison was made between PSK and VSB, it was apparent that for bandwidth efficiency, VSB had to be the choice. Furthermore, because VSB is relatively uncomplicated and therefore easy to implement, it again appeared that VSB should be the choice. However, when consideration is given to noise environment that may be present in unconditioned lines, the quadrature component distortion problem of VSB, and the requirement of pilot tones and restorer circuits for VSB, it was determined that the PSK technique, although more complex, had a better chance of success on this program, even with the bandwidth constraint. Furthermore, relatively little

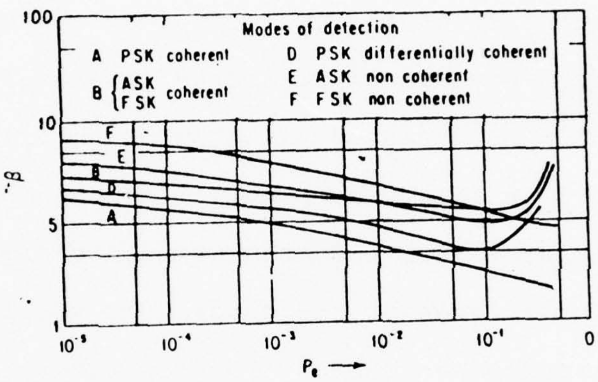


Figure 2.2.4-1 Comparison of binary systems on a β factor basis. Modes of detection: A, PSK coherent; B, ASK coherent and FSK coherent; D, PSK differentially coherent; E, ASK noncoherent; F, FSK noncoherent. (From Hancock, Proc. Natl. Electron. Conf.)

$$\beta = \frac{E_{min}}{No}$$

E_{min} = minimum received signal energy required per bit of information for a given error rate.

No = uniform gaussian noise-power spectral density of "No" watts/cycle for a given error rate.

data could be found during the research phase of the program on the performance of VSB. As a result, PSK was chosen over VSB for use as the modulation technique.

It should be noted, at this point, that in the above evaluation of the candidate modulation techniques, no information was provided on filter complexity which, of course, plays an important part on the efficiency of the modulation process. (Filtering here is in regard to the modulation process and should not be confused with the notch filter which was to be placed in the voice path.) The reason the filters were not considered for the modulation technique evaluation was because all three of the candidates required filtering in their processing schemes. Hence, they were all considered equal at this point.

In conclusion, as can be determined by the previous paragraphs, the PSK modulation technique was chosen as the means of translating the data into a suitable form for transmission over the wireline and within the 240 Hz slot. This conclusion was reached after careful study of available literature - some of which was presented in the preceding paragraphs - evaluating the various FAA requirements, and weighing the advantages and disadvantages of each technique with respect to its probability of success on this program. The circuits used for the modulator/demodulator are shown as part of the system schematic diagram Figures 2.5-1 and 2.5-3.

2.3 TECHNIQUE OF INSERTING A DATA SIGNAL INTO A VOICE CHANNEL

There are a number of alternate concepts for configuring a system in which both speech and data are to be exchanged between selected end point terminals. The most common of these concepts are listed below:

(a) Concept which employs a specific segment of the normal voice band for voice, while another segment is used for data transmission. Two popular techniques consistent with this concept are the slot-signalling and speech-plus techniques.

(b) Concept which employs separate transmission circuits for voice and data. The out-of-band signalling technique is an example of this concept.

(c) Concept which employs a time-sharing operational mode of the normal voice band for alternate voice/data transmission.

The investigation of this study focused on concept (a) above, i.e., utilization of a segment in the voice frequency band for the transmission of data. This concept was chosen primarily because it would have the best effect on speech quality; and with respect to concept (b) above, it was more economical when considering circuit implementation and cost. Furthermore, concept (b) would require, by its own nature, additional transmission media links. With respect to concept (c), this concept was not considered a good candidate for this study because the FAA considered push-to-talk (PTT) signals to be part of the data transmission scheme. Consequently, the PTT signal had to be transmitted simultaneously with the voice since it is under this condition that the PTT is required.

Since the slot signalling technique appears to offer the most promise, it is worthwhile to look at applications of the approach in other systems. Additionally, the other concepts (b and c) will be discussed to provide the FAA with some basic information for their own evaluation of the techniques. Included will be a summary and conclusion section which, again, will aid FAA personnel in formulating rationale with respect to the advantages and disadvantages of each concept.

2.3.1 SLOT SIGNALLING TECHNIQUE

The signalling arrangements employed in the SAGE airground voice communications system for establishing connections between Intercept Directors (IND), at the Direction Center (DC) and their assigned radios at a radio site, made use of a technique referred to as "slot signalling." Signalling was accomplished by use of the 43Al carrier telegraph terminal. The 43Al carrier terminal operates on a frequency shift basis and is regularly used in carrier telegraph systems. The operating frequency selected is considered as the nominal midband frequency, whereas the actual operating frequencies for signalling purposes are 35 cycles above and below this amount. The initial implementation was entirely on a point-to-point basis, employing two channels centered on midband frequencies of 2465 and 2635 Hz, respectively. The lower of the two frequencies, used for transmitting from the radio site to the Direction Center, and the higher frequency was used for transmitting from the Direction Center to the radio site. Signalling was accomplished by a frequency shift on the lower band between 2500 and 2430 Hz, and on the higher band between 2670 and 2600 cycles.

When the system was later modified for operation over the AUTOVON Switched Network, the frequencies of the 43A1 carrier needed to be changed. This was necessary to disable echo suppressors and to permit full duplex operation. In the Direction Center to radio site direction, the frequency shift from 2260 to 2330 Hz was employed to turn on the radio site transmitter. In the radio site to Direction Center direction, a frequency shift from 2090 to 2160 cycles provided an indication of receiver CODAN operation or a transmitter "carrier-on" indication.

The diagram shown below illustrates the use of the 43A1 channels in a 4-wire telephone circuit, with terminals bridged across the tip and ring of each pair at each end of the circuit. (For purposes of description, only the sending (S) and receiving (R) filter portions of the 43A1 are designated). Band rejection filters, shown in the line on the drop side of the terminal equipment, present a high loss to a particular frequency band. The purpose of these filters is to prevent the voice frequencies in the signalling range from modulating the 43A1 terminals, as well as to prevent the signalling frequencies in the speech spectrum from entering the trunk circuits.

From the arrangement shown in Figure 2.3-1 it can be seen that speech frequencies in the signalling range will not enter the trunk, and that signalling tones are confined to the trunks. Thus, signalling on a regular talking path was accomplished with only negligible interference with voice transmission due to filter action.

Pulse length code signals were used between the Direction Center and the radio site in order to establish correct channel connections.

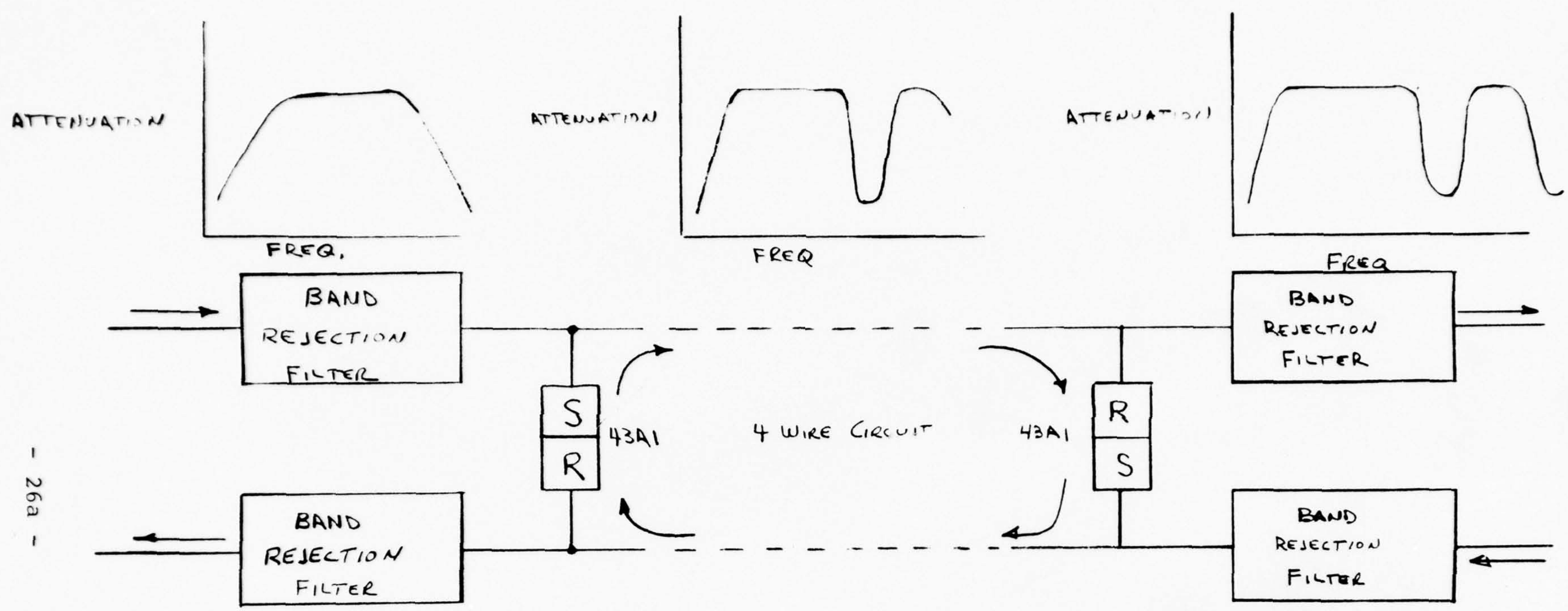


FIGURE 2.3-1 USE OF 43A1 TERMINAL WITH A 4 WIRE CIRCUIT

The code pulses originate in a code sender by the action of a pulse generator which provides uniformly spaced pulses at the rate of 25 pulses per second. At this rate, a single pulse uses approximately 20 milliseconds of pulsing time. A long pulse consisting of two short pulses bridged together is approximately 60 milliseconds. Pulses are accepted and terminated in a code receiver which registers the code and enables the link and controller to select the desired channel. A parity check in the code receiver ensures that the correct number of pulses are received in the proper sequence or no operation takes place. In such instances, the associated trunk circuits would time out and another trunk would be selected.

The trunk circuits employing the slot signalling technique experience a degradation of speech intelligibility since in the transmission and reception of speech, speech frequencies in the band blocked by the rejection filters do not reach the receiving unit. Under normal noise and operating conditions, the resultant impairment in intelligibility was determined to be equivalent to the insertion of an overall loss of about 2 dB in the transmission path. The location of the signalling band with respect to the frequency, required that consideration be given to the impairment penalty imposed by the band elimination. For this reason, the upper end of the speech band was used since theory indicates that the extraction from the voice channel of a band of frequencies of fixed width has a diminishing effect on the articulation impairment as the frequency of the band is increased, assuming that the noise is negligible or has about the same frequency spectrum as the speech.

To prevent interference between speech and signalling, the frequency components of speech in the signalling range were removed or blocked by the band filters. In a non-switch mode of operation, the system utilized WECO type 202E filters in the Direction Center to radio site direction, and WECO type 202F filters in the radio site to Direction Center direction. For switched mode of operation, the system employed WECO type 733A filters in the Direction Center to radio site direction, and WECO type 733B filters in the radio site to Direction Center direction. The discrimination at the signalling frequency was approximately 35 dB.

2.3.2 SPEECH PLUS TECHNIQUE

Another conventional technique, "speech-plus," is to install a low pass filter and impedance matching network splitting the voice channel into two segments: one for speech, and one for control tones. For example, on a 300 to 3000 KHz channel, a filter may be installed whose roll-off begins at 2600 Hz. Three tone channels could be placed above the filter. Using CCITT R35 frequencies, these would be 2700, 2820, and 2940 Hz. This limits the air-ground audio to a bandwidth of 300 to 2600, which generally provides an adequate speech bandwidth.

A typical application of "speech-plus" tones configured in such a manner has been used in a system installed by Intelect Inc., between Pago Pago and Apia in the Samoas. In this system, a 2600 Hz low pass filter is used. Transmitter keying is commanded via a 2700 Hz AM tone channel. Ground on the KY lead from the PTT switch sends a 2700 Hz tone over the channel. It is filtered and detected by the AM receiver and operates an output relay keying the selected transmitter. Main and standby transmitters are selected by a 2800 Hz frequency shift tone channel. The main transmitter is selected by keying the MARK frequency and the standby equipment is selected by keying the SPACE frequency.

AM tone channels provide tone-on for MARK and no-tone for SPACE. Frequency shift channels provide additional security by sending a low frequency for SPACE and shifting to a higher frequency for MARK.

The output relay of the 2800 frequency shift receiver is used to control the main/standby switching. This relay is an integral part of the tone receiver so an external relay module is not needed. A 2800 Hz frequency shift tone is also provided for report back status indication. The transmitter of this channel is located with the radio equipment and monitors the main/standby switching relay. The main/standby equipment status is then sent back to the equipment room and is used to directly operate the main and standby lamps on the selector switch. Illumination of either status lamp confirms that the selection command has been sent all the way to the transmitters and properly acted upon. Receivers are selected in the same manner using a 2900 Hz frequency shift channel for command and status report-back.

2.3.3 OUT-OF-BAND SIGNALLING TECHNIQUE

A technique applied in speech plus signalling circuits employs separate transmission paths for speech and signalling. In this type of configuration, two-wire or four-wire speech is completely isolated from the two-wire signalling signals. Transmission/reception of signalling information over the separate (out-of-band) signalling circuits is accomplished thru use of either single frequency AM tone sender/receiver equipment, or two-tone frequency shift keying equipment, much in the same manner as these signalling equipments are used in the "slot signalling" or "signal-plus" technique .

Since the signalling signals are physically separated from the voice signals, no special filtering is required in organizing the arrangement of the "out-of-band" termination equipments. However, the technique does require the use of a separate transmission circuit for each of the two information channels; namely, speech and signalling.

2.3.4 ALTERNATE SPEECH PLUS/DATA

This approach explores the popular technique of alternate voice/data communications employed in present day telecommunications systems.

Alternate VOICE/DATA arrangements permit the utilization of the same "VOICE" channel for transmission of VOICE signals or data information. In the conventional application of the alternate VOICE/DATA arrangement, a data call is placed in a similar manner to a regular voice call. To initiate a call, the attendant lifts the handset of the associated telephone set, receives a dial tone, and dials the telephone number of a distant telephone set associated with another alternate VOICE/DATA arrangement.

The call at the receiving end is answered in a normal manner. Upon distant end answering, attendants at both ends verify that the connection is ready to transmit and receive data. When modems at both ends are ready, the attendants operate their respective data keys and the modems may transmit and receive data.

If the call is to an automatic answering VOICE/DATA set, upon recognition of the answer signal, the attendant operates the data key and the modems may transmit and receive data. At the completion of transmission, the attendants normally return the data keys to their initial positions and hang up the handset.

Alternate VOICE/DATA permits the utilization of the total VOICE channel for transmission and reception of data during "VOICE-IDLE" periods, thereby permitting a faster data vote to be used for data information transfer.

Adoption of this technique would require certain procedural modifications for application to the specific requirements under consideration in this study.

In the first place, transmission circuits are of the non-switched category and, therefore, dial-up procedures would not be required. Secondly, data information transfer is generally machine-to-machine communications, without human attendance, therefore, automated means must be employed to detect "voice-idle" periods during which data transfer can occur. Thirdly, since push-to-talk (PTT) signals are of a higher order priority to that of any other data information transfer; and since this approach is a voice or data communication path but never a voice plus data communication path, it becomes necessary to incorporate the PTT element into the VOICE operation mode rather than the DATA operation mode.

To accommodate the constraints identified above, it would be necessary to organize an equipment system which would employ a speech plus technique for the voice plus PTT mode. Automatic detection of voice-idle periods could be achieved by means of monitoring the receive side of the transmission circuit via a VOX type detector and the transmit side via a "PTT-active" detector.

SECTION 2.3
2.3.5 SUMMARY AND CONCLUSIONS

Four (4) alternate approaches have been examined as potential candidates for implementation. In summary these are:

1) Slot Signalling
2) Speech Plus
3) Out-of-Band
4) Alternate Voice/Data

Design complexity of the slot signalling and speech-plus techniques appear to be of the same relative order, with perhaps a slightly lesser complication of filter design associated with the speech-plus technique. However, long term transmission service reliability appears to favor the slot-signalling technique. Practical concerns associated with sustaining a stable frequency bandwidth over a long service period to accommodate the upper band signalling portion of the speech-plus technique produce marginal assurances of service adequacy. Bandwidth truncation, a condition experienced in actual field service experience, is a potential problem in the speech-plus technique, when the signalling band borders the frequency spectrum which becomes affected by the bandwidth shrinkage.

Out-of-band signalling techniques offer a potentially highly reliable alternative; however, since this approach requires a seperate transmission facility for speech and another for signalling, it becomes economically prohibitive.

Alternate voice/data techniques offer a potentially desirable approach, where high data rates are required, since the total voice bandwidth is used during the data information transfer time. However, several operationally oriented problems become apparent with this technique. First, the technique is a time shared approach in which voice or data is capable of being transmitted, but not simultaneously. Second, it requires resolution of priority ordering. That is, since the approach precludes simultaneous voice and data transmission, preemptive algorithms must be applied to the operational mode, whereby one or the other (voice or data) information transfer mode will have an overriding capability. Additionally, the technique requires further investigation to resolving the unique requirement associated with the push-to-talk control feature of an Air Traffic Control System. Rapid alternate transfer from the voice to data mode becomes a technical constraint to the application of this technique for accommodating voice and push-to-talk transmission.

The speech-plus technique, theoretically, offers a potentially higher data transfer rate capability. (Assuming, of course, that a larger portion of voice bandwidth can be assigned to the data transfer function within the speech-plus concept than that for the slot-signalling concept.) However, this theoretical assumption may not be fully realized in practical applications because of the previously described bandwidth instabilities.

The out-of-band technique as well as the alternate voice/data technique are considered less desirable approaches; economics being a strong deterrent for the out-of-band technique, while operational difficulty and economics constraining the alternate voice/data concept.

Of the four techniques examined, the slot-signalling techniques offer the most promising technical/economical alternative. It offers relatively high speech quality; however, it is the most limiting technique from the standpoint of data transfer rates. Hence, its viability is dependent upon matching its data transfer rate capability with that of the system's operational needs.

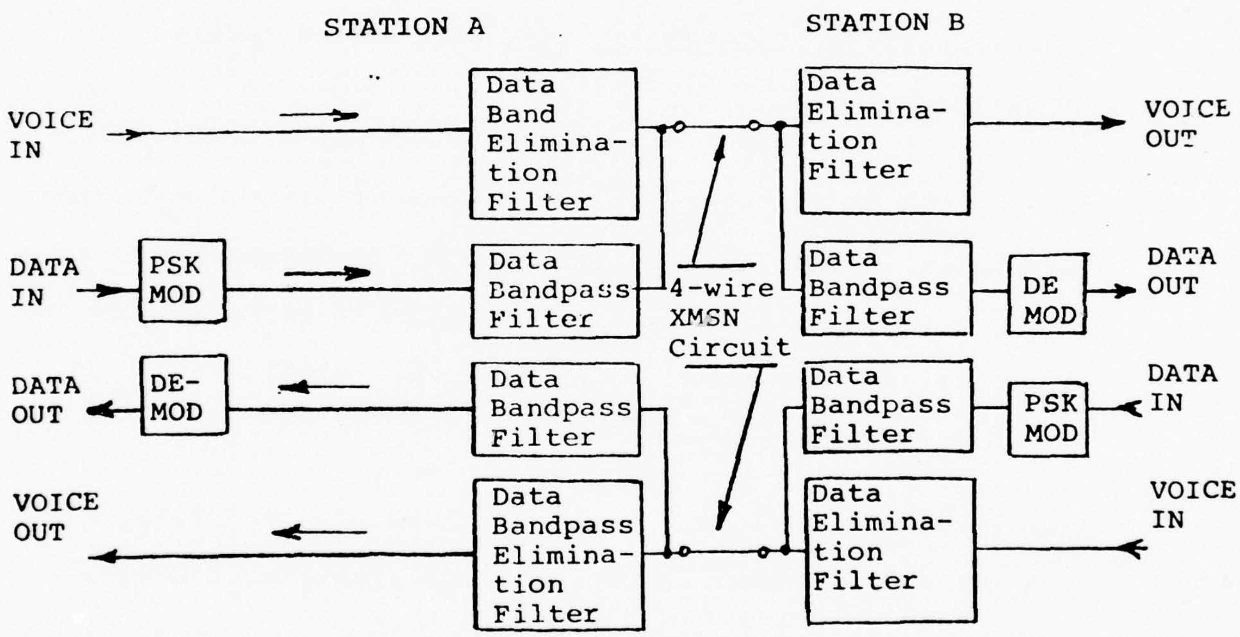
Through theoretical comparative analysis of various modulation techniques, it had been determined (ref. paragraph 2.2.4) to employ a Phase Shift Keying (PSK) modulation technique for the transmission of data within the data sub-channel element of the voice/data circuit.

This technique offers a superior approach for maximizing the data bit rate transfer within the least consuming bandwidth, extracted from the normal voice channel band as well as minimum error rates for a given signal-to-noise ratio.

Positioning of the data sub-channel was determined from reviewing CCITT and Bell System technical literature relating to in-band signalling, as well as the requirements established by FAA requirements for both speech quality and operational performance (ref. paragraph 2.1).

Based upon the composite constraints, and cautions derived from both of these sources, it was determined to position the data sub-channel within the frequency band limits of 2000 Hz and 2350 Hz.

Hence, the simplified block diagram of the arrived-at system arrangement is as illustrated below. The arrangement represents a full duplex voice/data channel in which both voice and data signals are to be transmitted either simultaneously or non-coincidentally.



Data band elimination filters are required in the voice transmit and receive paths to suppress data signals from entering into the voice channel. Similarly, data bandpass filters are required in the data transmit and receive paths to suppress voice signals from entering into the data sub-channel.

2.4 FILTER REQUIREMENTS

The filter requirements - a band reject or notch filter for the voice circuits and a bandpass filter for the data circuits - for the slot study program were primarily guided by the 240 Hz frequency restriction, i.e., pass all data signals through the slot at the highest possible speed without interfering with the voice circuits; and conversely, pass the maximum amount of voice signals without interfering with the slot data circuits. In order to meet these goals, several iterations of filter networks were investigated. Upon selection of a network, a filter was designed, constructed and tested to determine its operational performance within the constraints of the slot system.

2.4.1 TWIN-T AND MULTIPATH FILTERS (First Iteration)

The first approach considered to meeting the filter requirements was to incorporate a Twin-T network for the voice notch filter on a multiple feedback network for the data bandpass network. The Twin-T was considered for the voice notch because a high-Q or sharpness of the filter roll-off characteristic could be obtained, thus providing the system with the narrow bandwidth required and a high rejection at the notch center frequency. The circuit designed (Ref. Figure 2.4.1-1) consisted of a positive unity gain amplifier, a RC Twin-T network (notch frequency control) and a RC-T network (Q control).

NOTCH FILTER DESIGN
(TWIN - T TYPE)

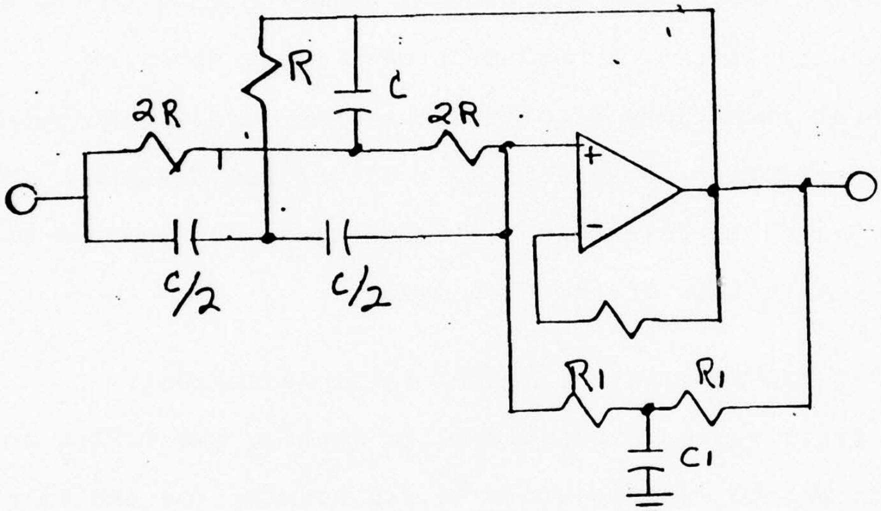
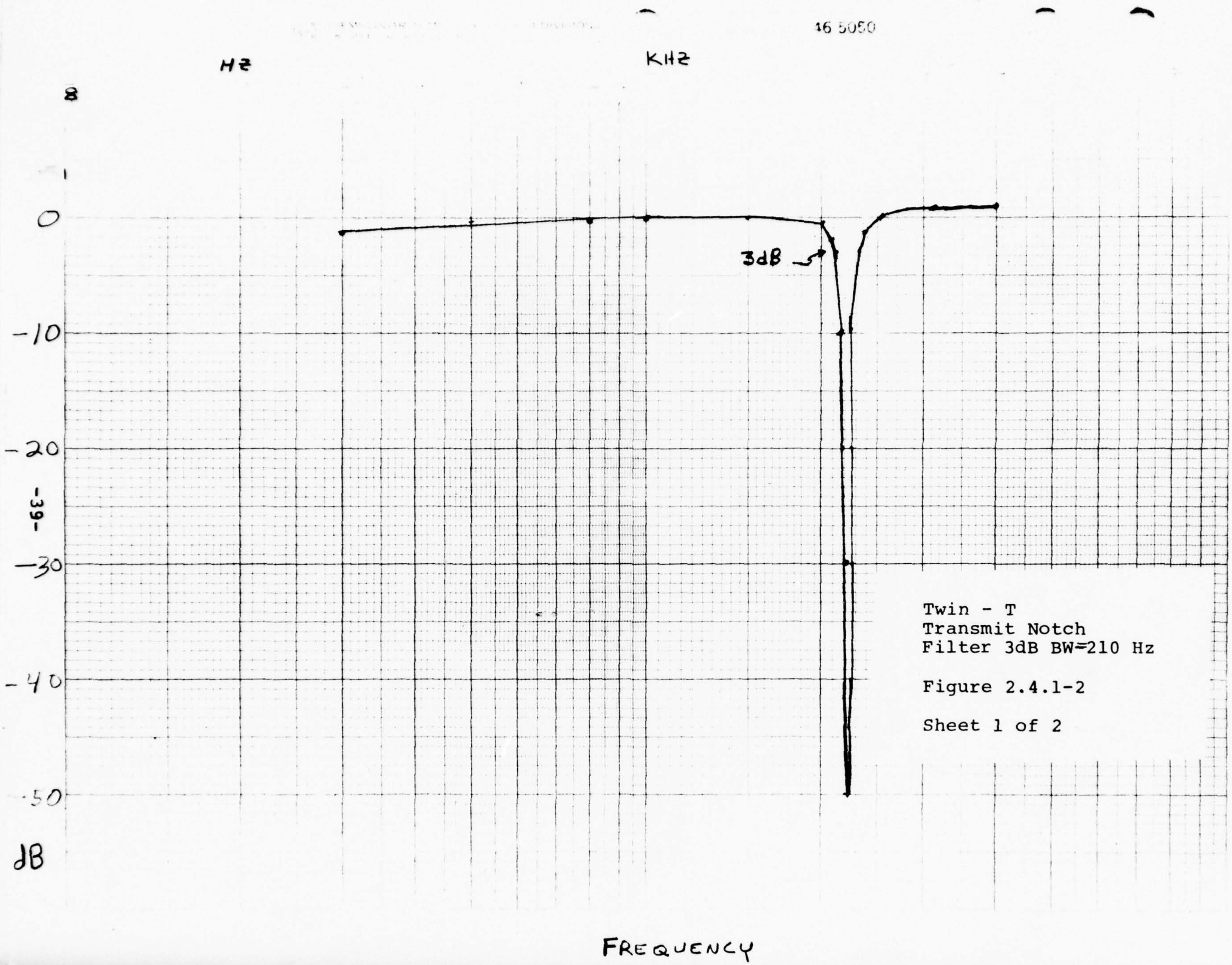


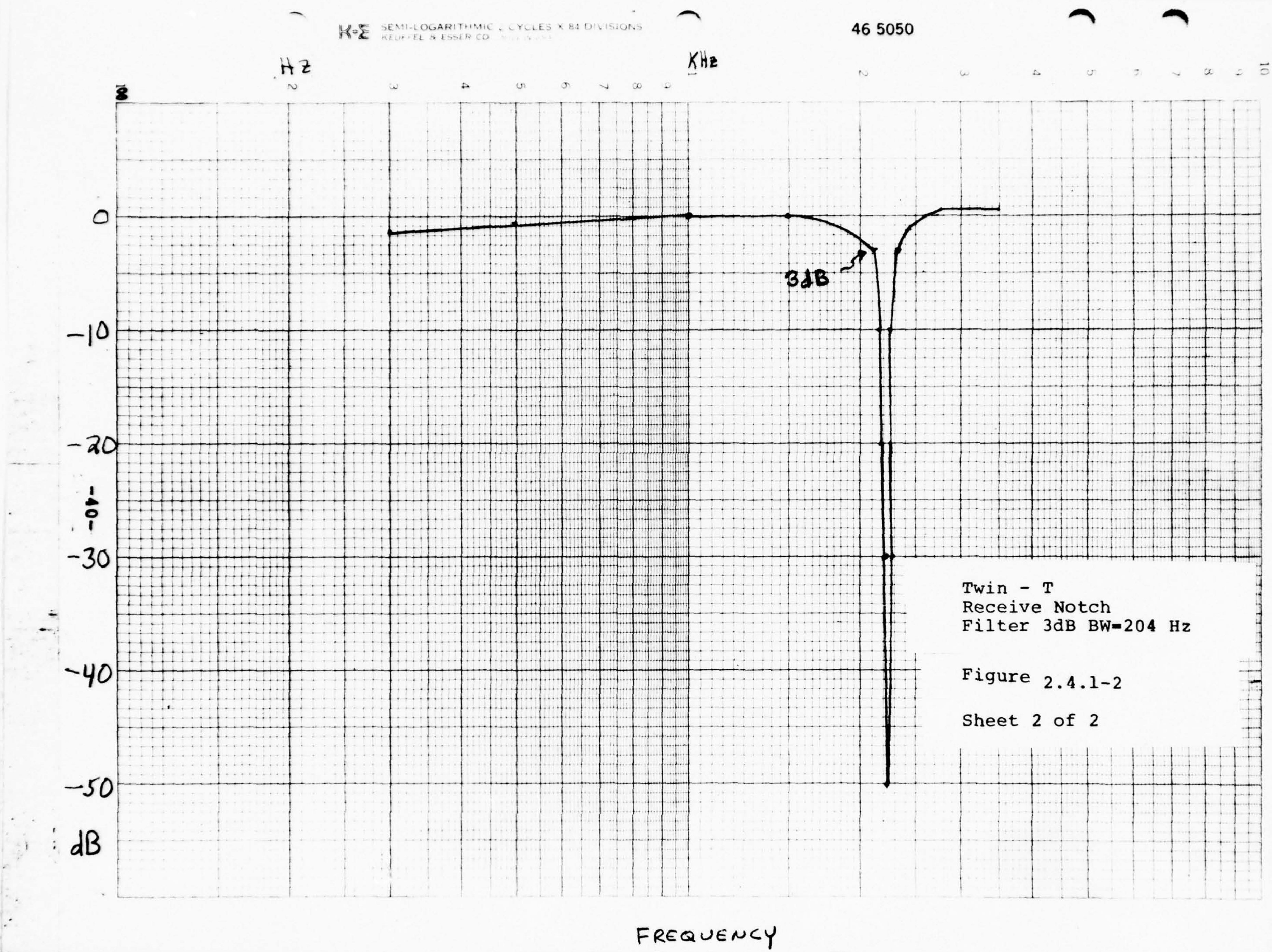
FIGURE 2.4.1-1



Twin - T
Transmit Notch
Filter 3dB BW=210 Hz

Figure 2.4.1-2

Sheet 1 of 2



Twin - T
Receive Notch
Filter 3dB BW=204 Hz

Figure 2.4.1-2

Sheet 2 of 2

BAND PASS FILTER DESIGN

(MULTIPATH TYPE)

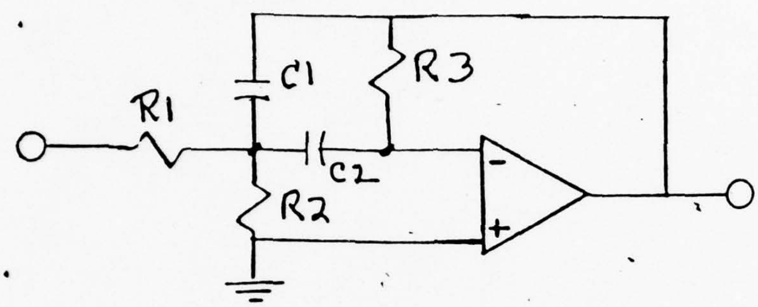
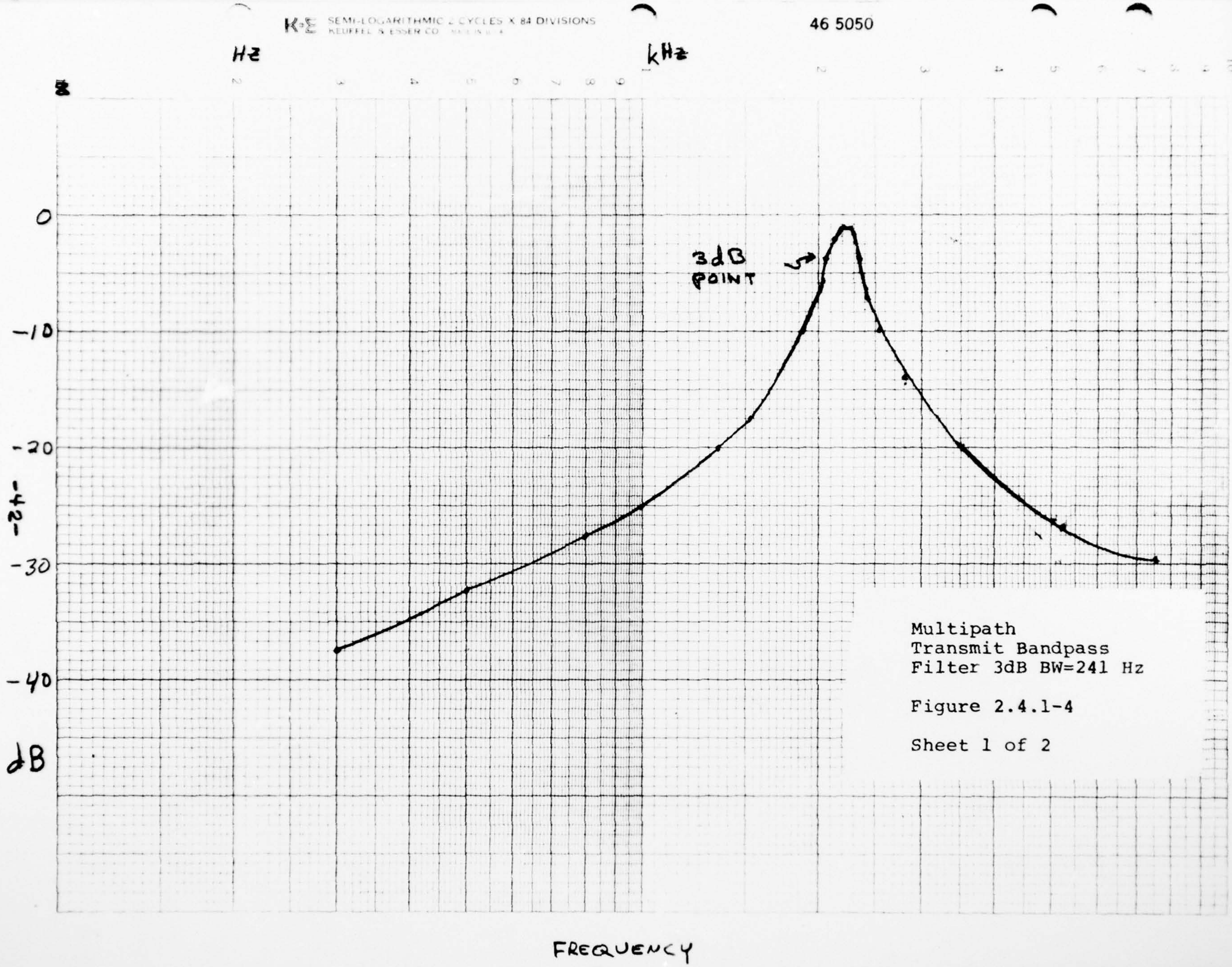


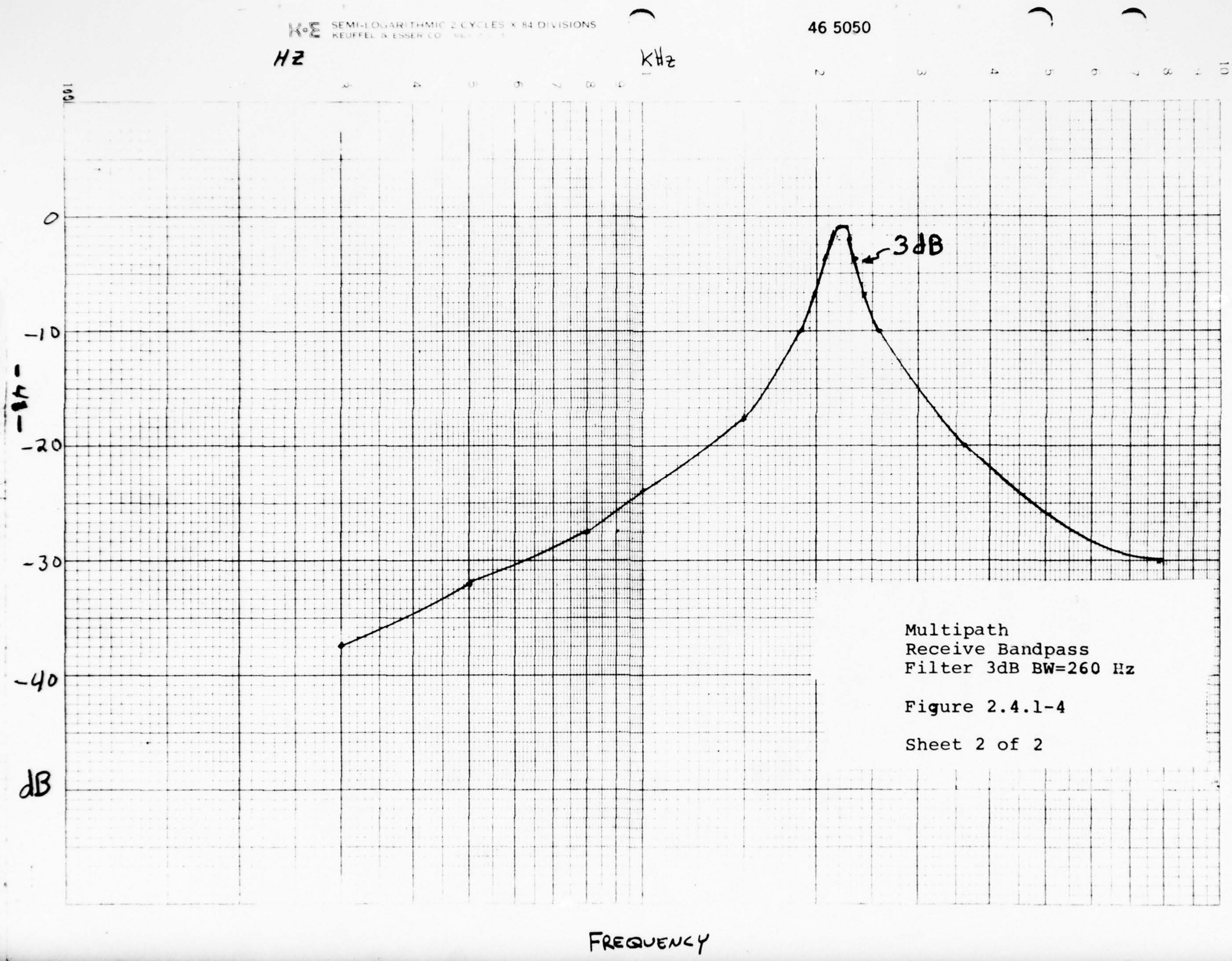
FIGURE 2.4.1-3



Multipath
Transmit Bandpass
Filter 3dB BW=241 Hz

Figure 2.4.1-4

Sheet 1 of 2



Multipath
Receive Bandpass
Filter 3dB BW=260 Hz

Figure 2.4.1-4

Sheet 2 of 2

The multipath-feedback network was chosen for the data band-pass filter because of its lack of phase shift sensitivity. (Ref. Figure 2.4.1-3). The filter frequency response characteristics of each of the networks (Twin-T and multiple feedback) are shown in Figure 2.4.1-2 and 2.4.1-4, respectively. The key characteristics of the filters are summarized below:

Filter	Type	B W (3dB)	Filter Depth
Band reject (notch) (Transmit)	Twin T	210 Hz	@ 2210 Hz 50 dB
Bandpass (Transmit)	Multiple feedback	241 Hz	Center frequency tuned for 2210 Hz
Band reject (notch) (Receive)	Twin T	204 Hz	50 dB
Band pass (Receive)	Multiple feedback	260 Hz	Center frequency tune for 2210 Hz

The above filter circuits were installed in the test configuration described in section 2.6.

Voice quality measurements were determined to be of a satisfactory nature "under no data transmitted" conditions; however, when data was transmitted along with the voice, mutual interference of voice and data were observed in each respective signal channel. The problem encountered can be summarized as follows:

The filters were designed and constructed to be of approximately comparable bandwidths. Hence, although the bandwidths were comparable at the 3 dB points, an overlap of frequencies was observed. The rejection of the notch filter was not broad enough to eliminate all the data modulation products, therefore, some of these products were heard in the voice circuits; and conversely, the bandpass filter's bandwidth was broad enough to accept signals from the voice path, thereby decoding these signals as if they were data signals and causing error measurements to be made in the data circuits.

As a result of this test, it became apparent that more complex filter types would have to be incorporated which would provide more coincidental type filters, i.e., the filters must approach theoretical perfect filters. The band reject must have a broader bandwidth at the 30 or 40 dB points. Additionally, the phase characteristics of both the band reject and the bandpass filters must be improved. The improved phase characteristic would aid in reducing the effect of harmonics which may be generated by the bandpass filter in the transmit section, and were not being completely eliminated by the band reject filter in the receive section.

2.1.2 4TH ORDER FILTERS (Second Iteration)

In order to provide filters with improved characteristics, Orion Engineers contacted vendors who specialize in these filter designs. Two significant vendors contacted were Burr-Brown and National Semiconductor, both of whom provide filter IC's. Mr. George Warren of National Semiconductor was most helpful, and the problem encountered on the first iteration was explained to him. With the information provided to him, he proposed a design utilizing their AF 100 Universal Filter IC, and ran a computer program which provided improved attenuation and phase characteristics as well as defining the pole and zero locations for the respective filters. This information, as well as the frequency response curves and circuit designs, are provided in Figures 2.4.2-1 thru 2.4.2-8. The characteristics of the filters are tabulated below:

Filter	Type	BW (3 dB)	Filter Depth
Band reject (notch) Transmit	4th Order (Butterworth)	237 Hz	48 dB @ 2210 Hz
Band reject (notch) Receive	4th Order (Butterworth)	172 Hz	45 dB @ 2210 Hz
Bandpass (Transmit)	4th Order (Elliplical)	230 Hz	Tuned for pole locations of 2123 Hz and 2293 Hz (both sections)
Bandpass (Receive)	4th Order (Elliplical)	241 Hz	

Measurements utilizing this newly constructed filter, although exhibiting problems similar to those experienced in the initial iteration, did indicate improvement in transmission; i.e., low speed data rate, less than 50 bps, was accomplished, but, again, the narrow bandwidth of the receive band reject filter still allowed data tones in the voice path and some data errors were still encountered in the data path. Additionally, the second iteration essentially proved what was found as a result of the first iteration, i.e., the original objective established

AF 100 TYPE FILTER
4th ORDER DESIGN INFO
NOTCH (BUTTERWORTH)

TRANSFORMED POLE/ZERO LOCATIONS
FILTER # 1

FILTER CENTER FREQ. 2206.7 BANDWIDTH 240

POLE LOCATIONS
CENTER FREQ. Q

2123.43 13.01076
2293.2354 13.01076

JW AXIS ZEROS

2206.7
2206.7

FIGURE 2.4.2-1

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NORMALIZED AT FREQUENCY 1

FREQUENCY	GAIN	PHASE	DELAY
2	-.38577424	-48.320435	.75659297
2.01	-.4706879	-51.191114	.84015402
2.02	-.57862246	-54.384377	.93607619
2.03	-.71685066	-57.947604	1.0459423
2.04	-.89513312	-61.933525	1.1710904
2.05	-1.1265266	-66.399658	1.3121984
2.06	-1.4283495	-71.399826	1.4686263
2.07	-1.82322	-76.997828	1.6374935
2.08	-2.3399615	-83.197457	1.8126182
2.09	-3.0140356	-90.034028	1.9837317
2.1	-3.8871186	-97.458676	2.13674
2.11	-5.0057206	-105.37782	2.2558853
2.12	-6.4195705	-113.64422	2.3279229
2.13	-8.1817392	-122.0747	2.3468354
2.14	-10.353589	-130.48191	2.3164411
2.15	-13.018429	-138.70864	2.2491566
2.16	-16.310057	-146.65173	2.1617649
2.17	-20.472724	-154.26917	2.0707748
2.18	-26.010734	-161.57305	1.9994257
2.19	-34.193105	-168.61558	1.926736
2.2	-50.096763	-175.47426	1.8878731
2.21	-62.438554	177.76052	1.8749465
2.22	-38.264891	170.99529	1.8876389
2.23	-28.569127	164.1417	1.9234363
2.24	-22.42332	157.12456	1.977469
2.25	-17.945581	149.891	2.0421696
2.26	-14.462539	142.42052	2.1071423
2.27	-11.662332	134.73435	2.1597833
2.28	-9.3782832	126.90058	2.1870963
2.29	-7.5103935	119.03062	2.1755059
2.3	-5.9903203	111.26489	2.1299701
2.31	-4.764303	103.75026	2.0399539
2.32	-3.7853683	96.616126	1.9191328
2.33	-3.010755	89.957338	1.7777197
2.34	-2.4018928	83.827724	1.626922
2.35	-1.9251073	78.243332	1.4751678
2.36	-1.5520754	73.191177	1.3321044
2.37	-1.2597491	68.639326	1.1966811
2.38	-1.0298489	64.545661	1.0777195
2.39	-.8481382	60.864318	.96958537
2.4	-.70365784	57.549854	.87376329

DELAY AT 1 = 2.1261397E-02

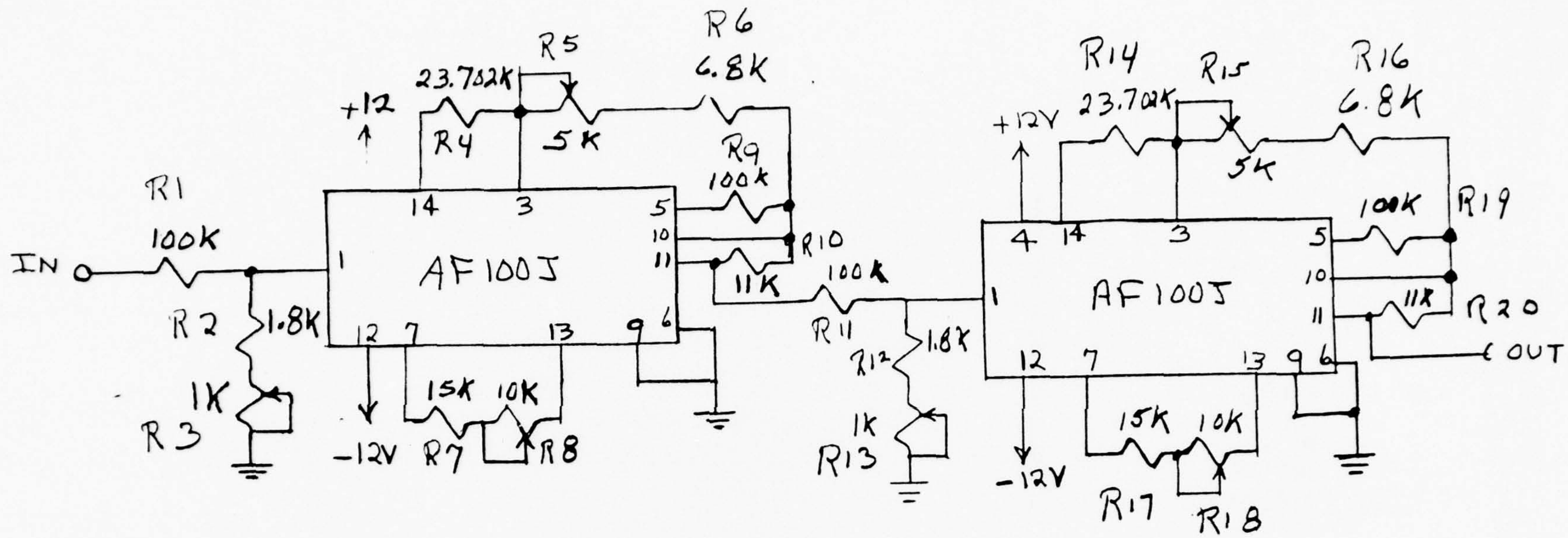
REAL POLE

NUMERATOR
A(1)= 1   B(1)= 0   Z(1)= 2.2067
A(2)= 1   B(2)= 0   Z(2)= 2.2067

DENOMENATOR
P(1)= 2.12343   Q(1)= 13.01
P(2)= 2.2932   Q(2)= 13.01

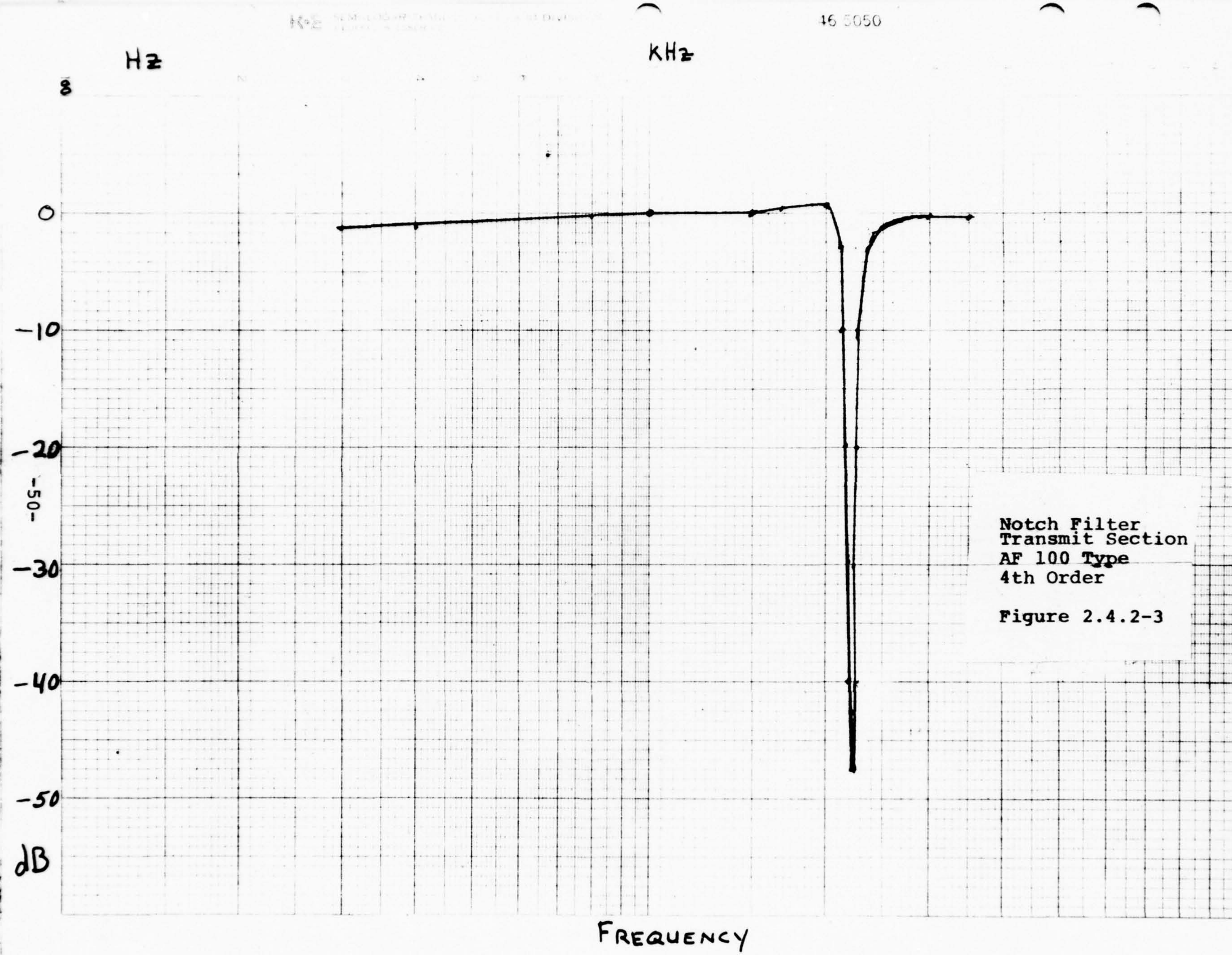
WHAT OPTION ?

FIGURE 2.4.2-1
SHEET 2 OF 2



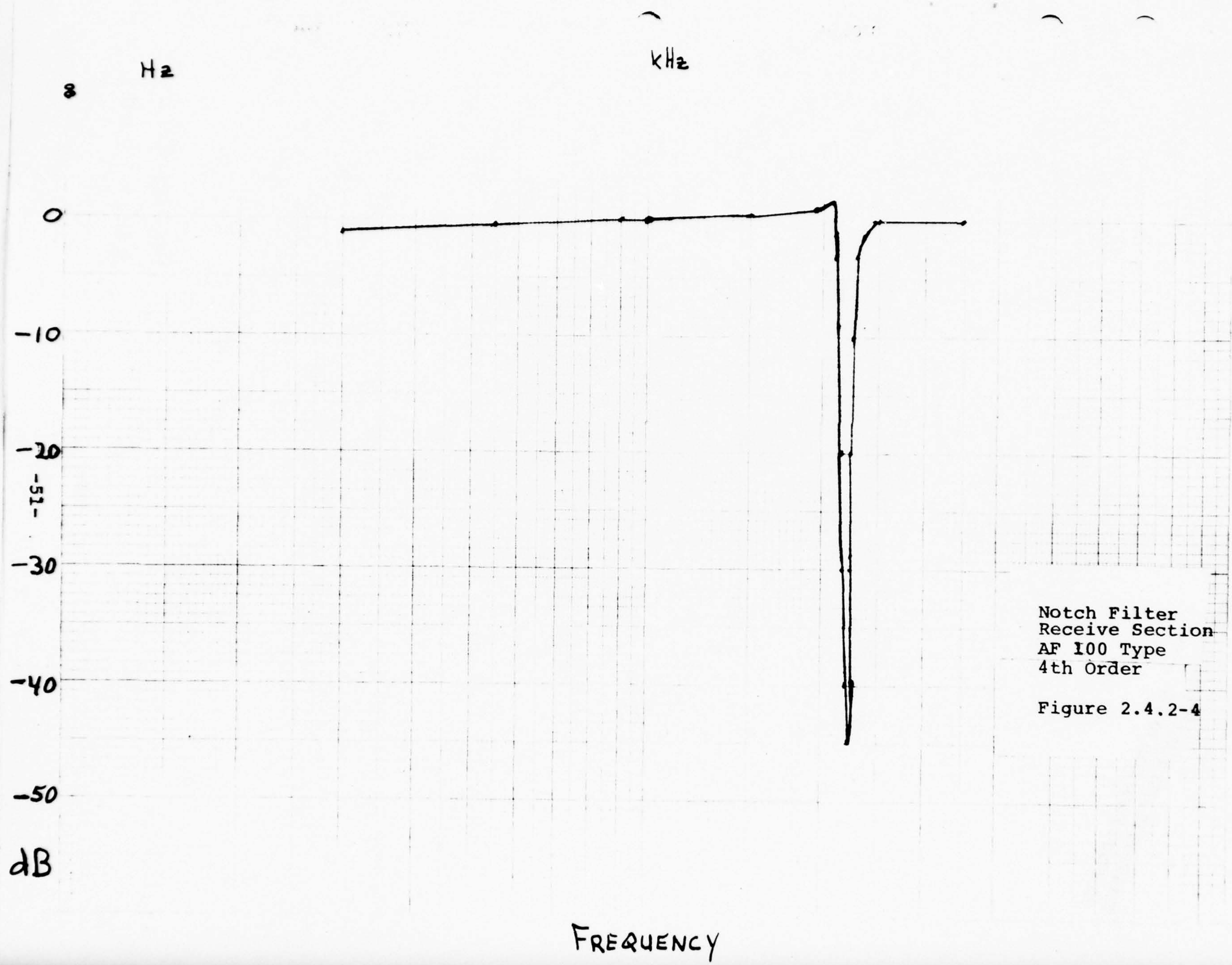
NOTCH FILTER (BUTTERWORTH)
4th ORDER

FIGURE 2.4.2-2



Notch Filter
Transmit Section
AF 100 Type
4th Order

Figure 2.4.2-3



Notch Filter
Receive Section
AF 100 Type
4th Order

Figure 2.4.2-4

AF 100 TYPE FILTERS
4th ORDER DESIGN INFO

BANDPASS (ELLIPTICAL)

TRANSFORMED POLE/ZERO LOCATIONS
FILTER # 1

FILTER CENTER FREQ. 2206.7 BANDWIDTH 240

POLE LOCATIONS
CENTER FREQ. 0

[illegible] 13.01075
[illegible] 13.01075

FIGURE 2.4.2-5
SHEET 1 OF 2

NORMALIZED AT FREQUENCY 2206.7

FREQUENCY	GAIN	PHASE	DELAY
1000	-48.297938	174.96718	2.1259824E-05
1100	-45.672963	174.14358	2.4653205E-05
1200	-43.033394	173.17855	2.9135633E-05
1300	-40.328734	172.02229	3.5389969E-05
1400	-37.50012	170.59995	4.4160214E-05
1500	-34.478401	168.79335	5.7093156E-05
1600	-31.171048	166.40353	7.727372E-05
1700	-27.443975	163.06652	1.1132669E-04
1800	-23.034674	158.03468	1.757217E-04
1900	-17.723995	149.43804	3.2109177E-04
2000	-10.708015	131.68364	7.5653397E-04
2100	-2.2886739	83.547133	2.1367243E-03
2200	-5.0570933E-05	4.3353682	1.887526E-03
2300	-1.2613831	-68.71379	2.1240709E-03
2400	-8.2095347	-122.44316	8.7403317E-04
2500	-14.427852	-142.94024	3.6193679E-04
2600	-19.306997	-153.34759	1.9659723E-04
2700	-22.915465	-157.80946	1.195489E-04
2800	-25.[illegible]	-161.36382	8.1356147E-05
2900	-28.[illegible]	-163.9695	5.9958151E-05
3000	-30.35079	-165.73313	4.5213119E-05
3100	-32.172659	-167.17916	3.5643305E-05
3200	-33.79743	-168.33888	2.8869695E-05
3300	-35.237571	-169.27919	2.3989685E-05
3400	-36.551623	-170.06735	2.012036E-05
3500	-37.752965	-170.73696	1.7196826E-05
3600	-38.[illegible]07	-171.31283	1.4898268E-05
3700	-39.[illegible]35	-171.81396	1.3017499E-05
3800	-40.93742	-172.25415	1.1499294E-05
3900	-41.7304	-172.64443	1.0281495E-05
4000	-42.[illegible]	-172.99302	9.1693229E-06

DELAY AT 2206.7 = 1.8753436E-03

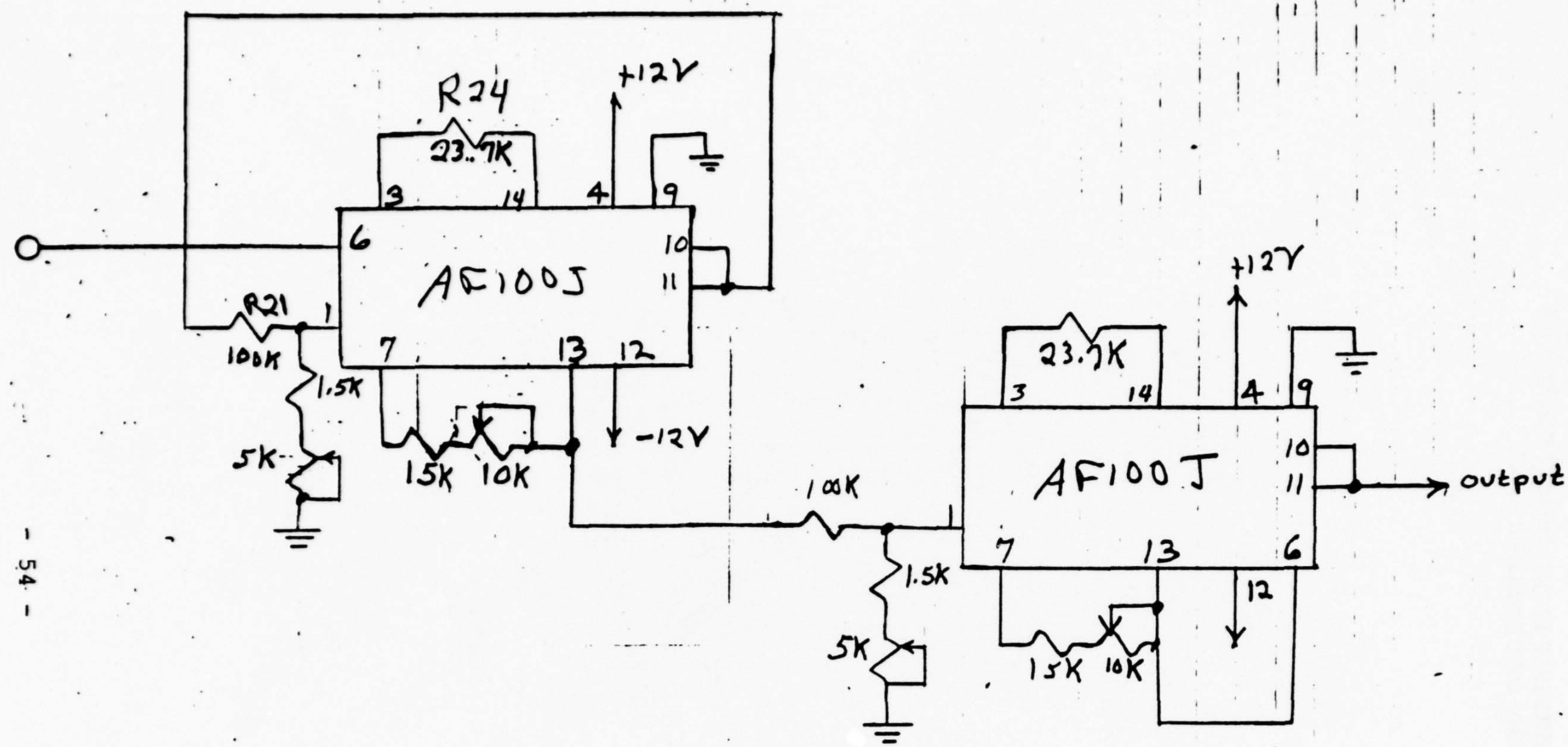
[illegible]

[illegible]
[illegible] 1)= 0 [illegible] 1)= 1 [illegible] 1)= 0
[illegible] 2)= 0 B(2)= 1 [illegible] 2)= 0

[illegible]
[illegible] 1)= -173.4[illegible] [illegible] 1)= 13.01076
[illegible] 2)= [illegible] [illegible] 2)= 13.01076

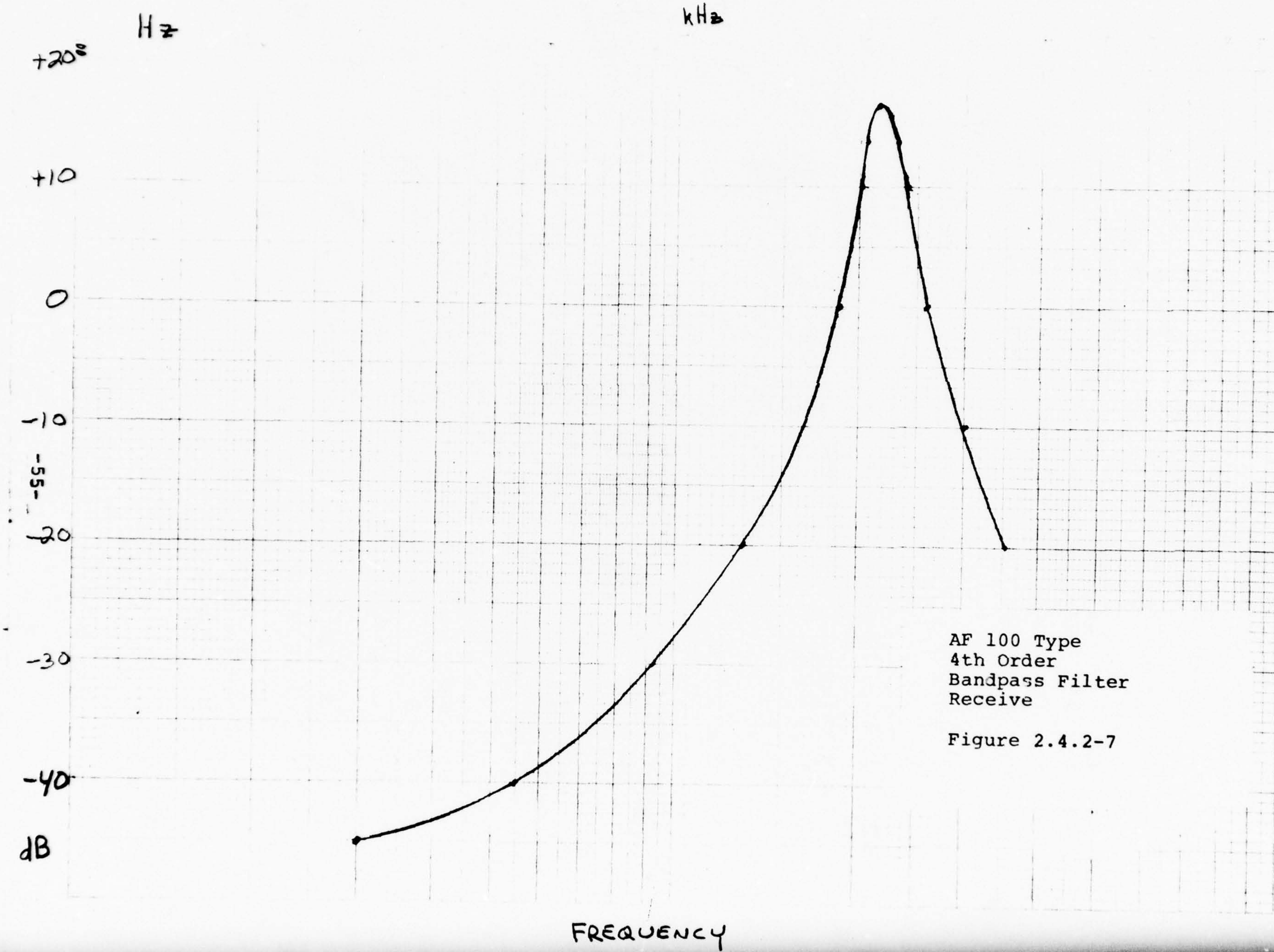
[illegible]
[illegible]

FIGURE 2.4.2-5
SHEET 2 OF 2



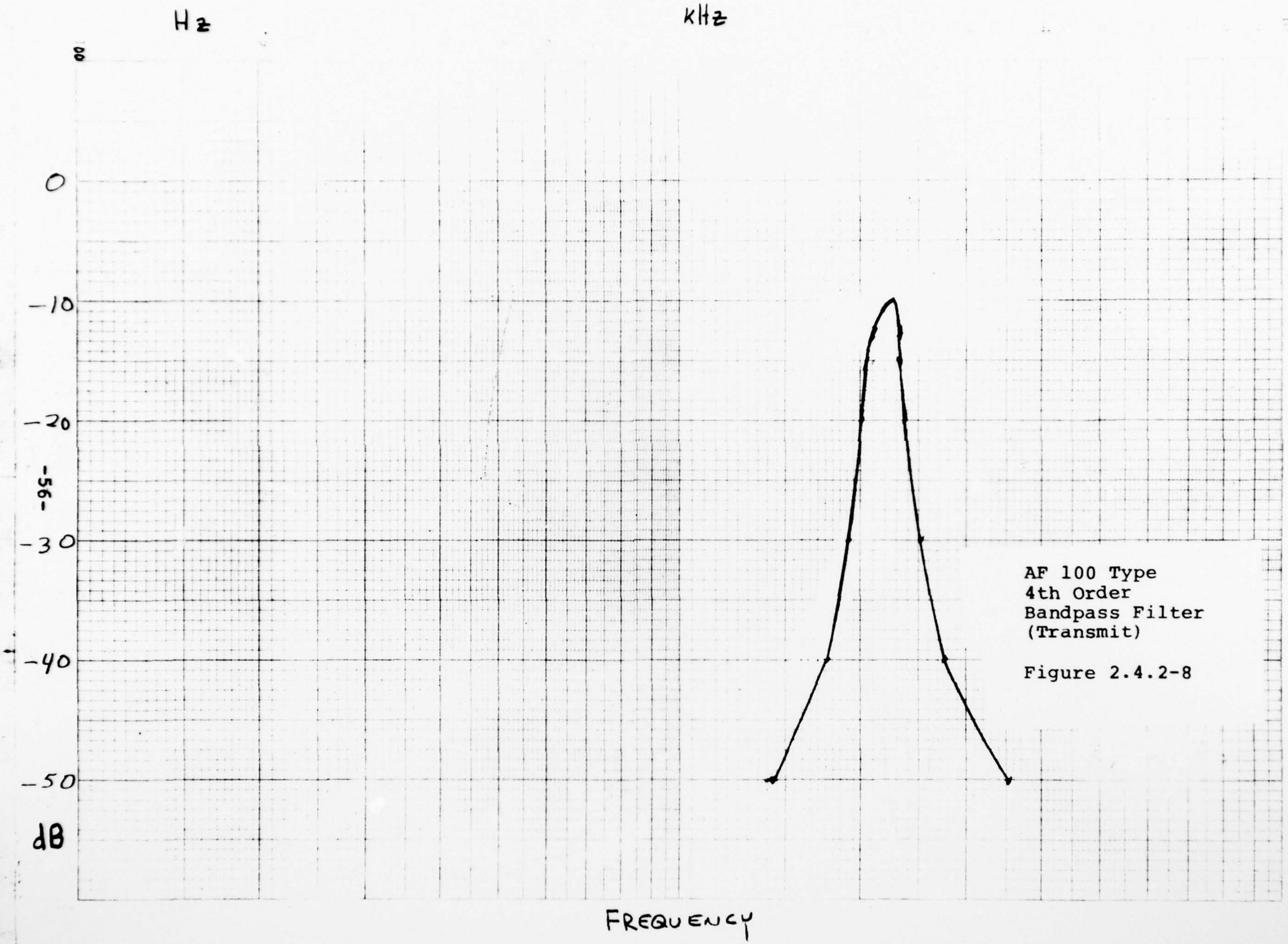
BANDPASS FILTER (ELLIPTICAL)
4th ORDER
(FOR TRANSMIT AND RECEIVE SECTIONS)

FIGURE 2.4.2-6



AF 100 Type
4th Order
Bandpass Filter
Receive

Figure 2.4.2-7



AF 100 Type
4th Order
Bandpass Filter
(Transmit)

Figure 2.4.2-8

for this study of "coincidental" bandwidth filters was technically not feasible. A 240 Hz notch and 240 Hz bandpass would not provide acceptable performance unless ideal filters could be used.

2.4.3 8TH ORDER NOTCH, 4TH ORDER BANDPASS (Third Iteration)

Having established a baseline, which indicated that the voice notch had to incorporate a broader bandwidth than the data bandpass in order to restrict mutual interference in the voice and data channels, the third iteration focused on broadening the voice notch bandwidth.

Additionally, it was observed that the receive data bandpass filter should also incorporate a broader bandwidth, but remain narrower than the notch, in order to accept practical data rates for operational performance. In order to meet these requirements of increased bandwidth, a more complex design to the notch filter was used, while retuning of the bandpass was all that was required. Mr. George Warren of National Semiconductor was again contacted and a new filter design proposed. This filter network would be of an 8th order design. Mr. Warren ran a computer program, utilizing essentially the same requirements of the 4th order design, but reidentifying and adding more poles and zeros to the network. With these new parameters identified, Orion engineers then designed and constructed the new 8th order notch filters with the circuits and frequency response as illustrated in Figures 2.4.3-1 thru 2.4.3-9 Additionally, the brief computer run identifying the pole and zero locations are included in Figure 2.4.3-1.

The key characteristics of the new filter are tabulated below

Filter	Type	BW (3 dB)	BW (30 dB)
Band reject (notch) (Transmit)	8th Order (Elliptical)	394 Hz	266 Hz
Band reject (notch) (Receive)	8th Order (Elliptical)	395 Hz	257 Hz
Bandpass (Transmit)	4th Order (Elliptical)	228 Hz	1050 Hz
Bandpass (Receive)	4th Order (Elliptical)	322 Hz	1050 Hz

Under this configuration a practical data bit rate of 200 bits per second, meeting the 1×10^{-4} error rate efficiency requirement, was achieved while transmitting both voice and data signals simultaneously. However, in order to achieve this efficiency rate, which still include modulation frequency products in the voice path, the voice signals had to be transmitted at a relatively low level (highest voice peak of -10VU), while the data level was at 0 dBm (600 ohms). This low voice level transmission was required because the filter characteristics were still not optimized, but to continue with iterations was beyond the scope of the program since it had already been determined that it was not feasible

```
LOWPASS FILTERS
WHAT TYPE OF FILTER ?  B-C-E
-2
 DO YOU KNOW THE ORDER OF THE FILTER ?  Y/N
? Y
INPUT FC,FS,AMAX,N
? 1,2,.1,4
     FC                1.000
     FS                2.000
     AMAX               .100
     AMIN            -41.447
     N                 4.000
     ATT AT FS       -41.447
IS THIS SATISFACTORY ?  Y/N
? Y

    F           Q
    1.133       2.630
     .858        .640
    Z
    4.922
    2.143

 DELETE ALL

 LOAD TRANS
RUN
 WHAT TYPE FILTER IS REQUIRED HIGHPASS, BANDPASS, NOTCH
? N
IS INPUT TO BE FROM FILE OR TEL
? FILE

 ENTER FREQUENCY SCALING FACTOR
? 1

 ENTER THE # OF FILTERS TO BE DESIGNED
? 1
 ENTER THE C.F. AND BW OF EACH FILTER
? 2206.7,480




                 TRANSFORMED POLE/ZERO LOCATIONS
                            FILTER # 1

 FILTER CENTER FREQ.      2206.7                  BANDWIDTH       480

          POLE LOCATIONS
   CENTER FREQ.              Q


 2009.2332       27.63087
 2423.5665       27.63087
 2033.236        5.065515
 2394.0879       5.065515
        JW AXIS ZEROS

 2158.4791
 2255.9982
 2047.5569
 2321.5222
```

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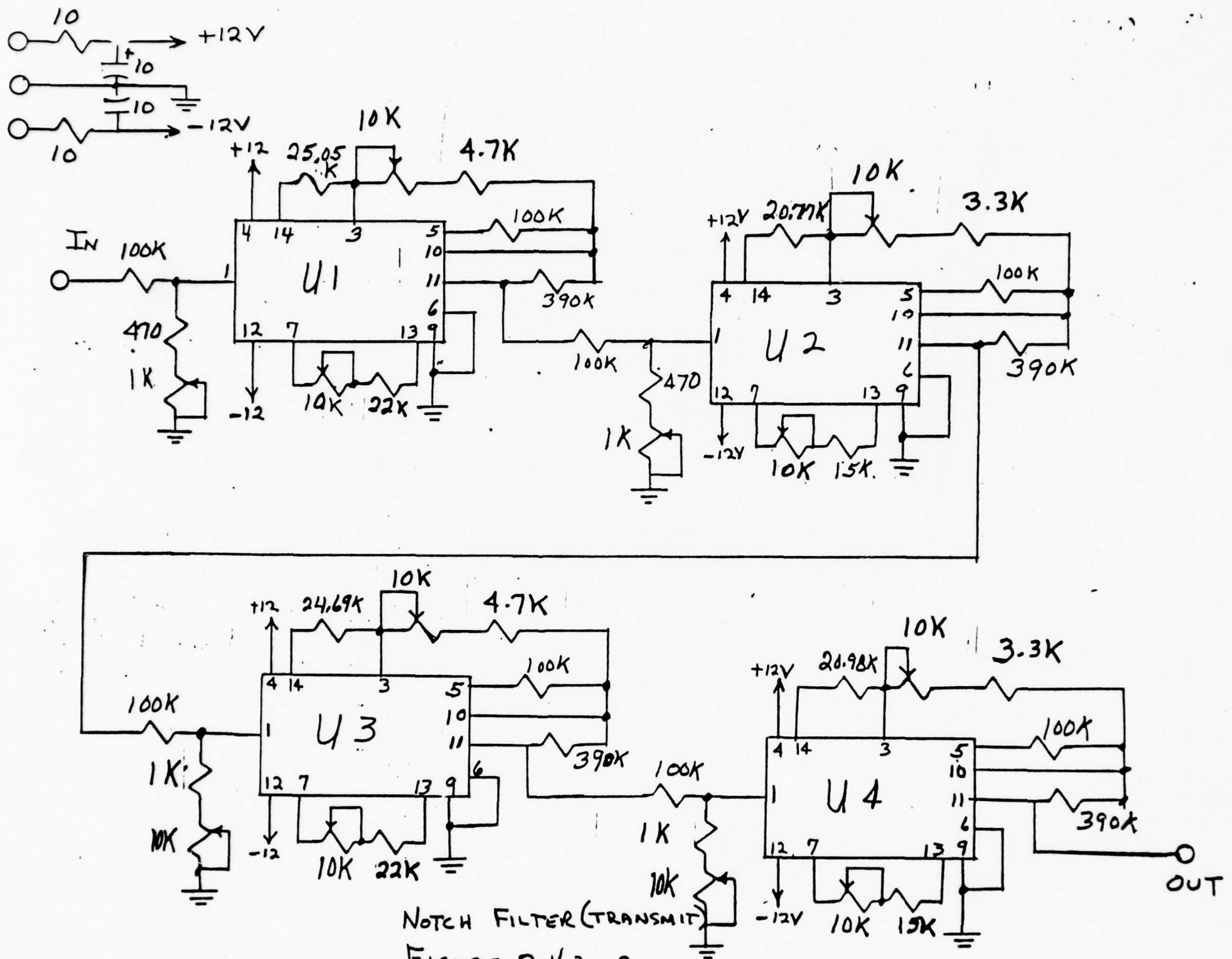
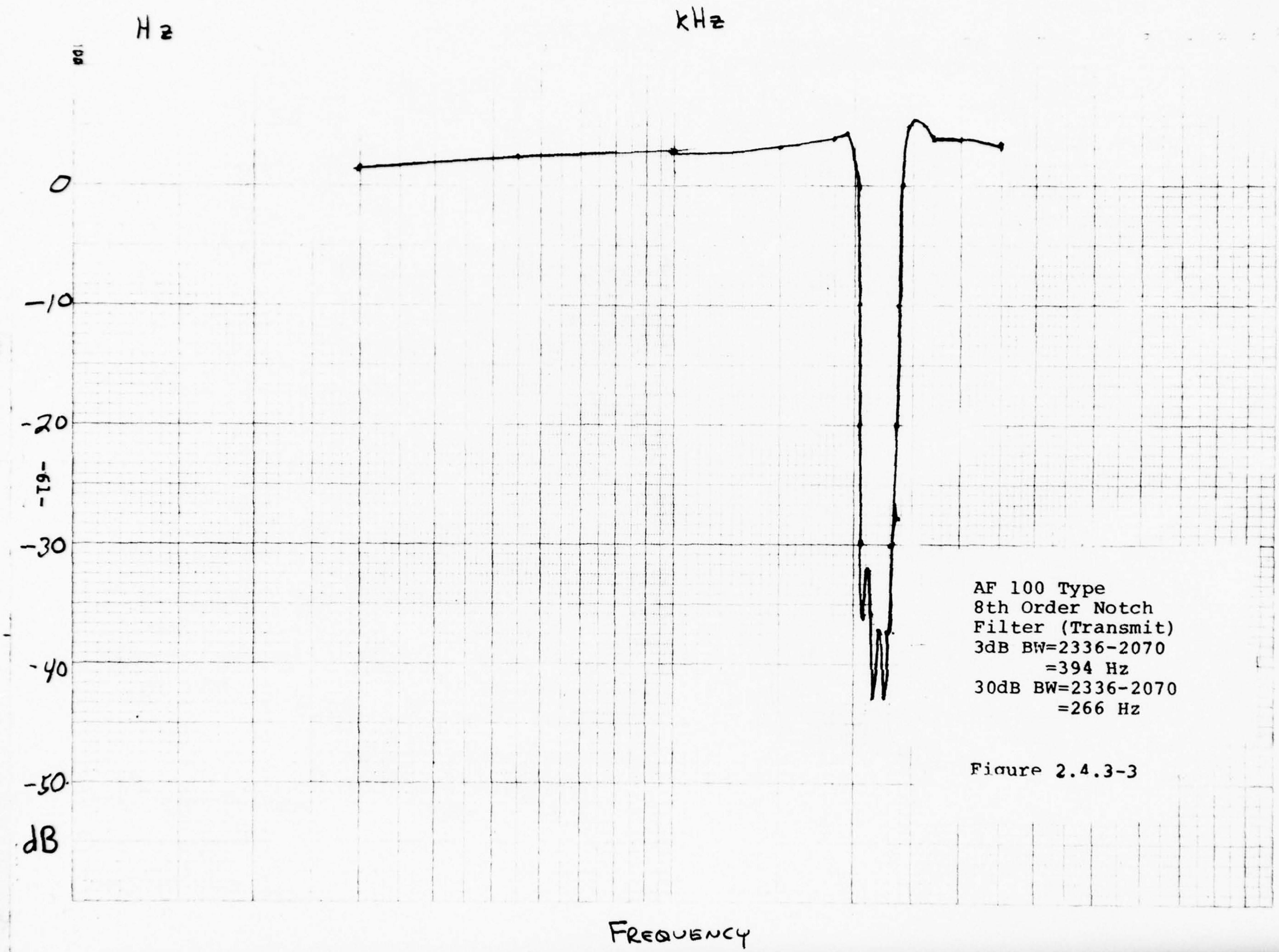
AF 100 TYPE
8th ORDER
(ELLIPITICAL FILTER)

FIGURE 2.4.3-1



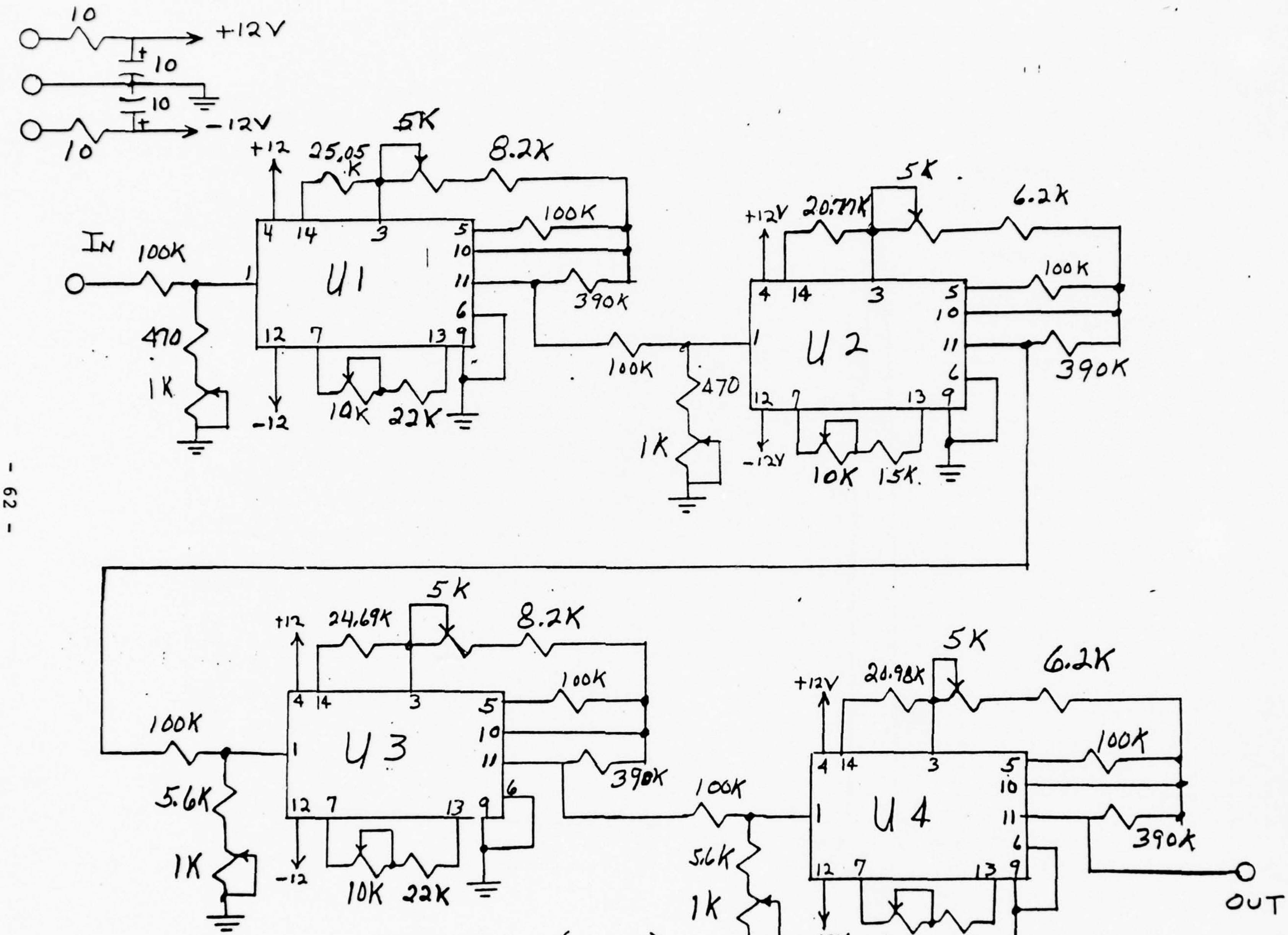
NOTCH FILTER (TRANSMIT)

FIGURE 2.4.3.-2



AF 100 Type
8th Order Notch
Filter (Transmit)
3dB BW=2336-2070
=394 Hz
30dB BW=2336-2070
=266 Hz

Figure 2.4.3-3



NOTCH FILTER (RECEIVE)

FIGURE 2.4.3-4

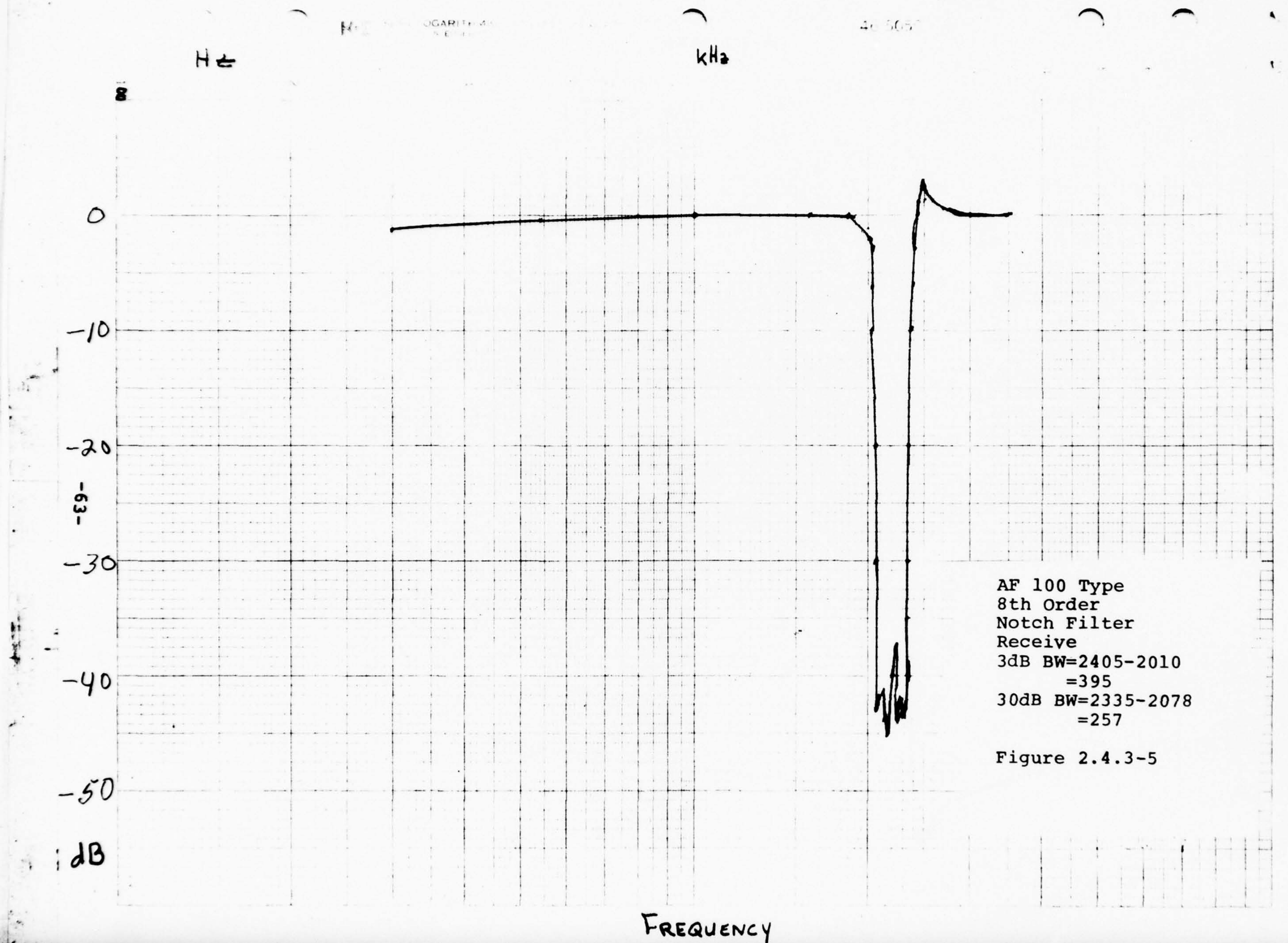

Hz
kHz
0
-10
-20
-30
-40
-50
dB
FREQUENCY
AF 100 Type
8th Order
Notch Filter
Receive
3dB BW=2405-2010
=395
30dB BW=2335-2078
=257

Figure 2.4.3-5

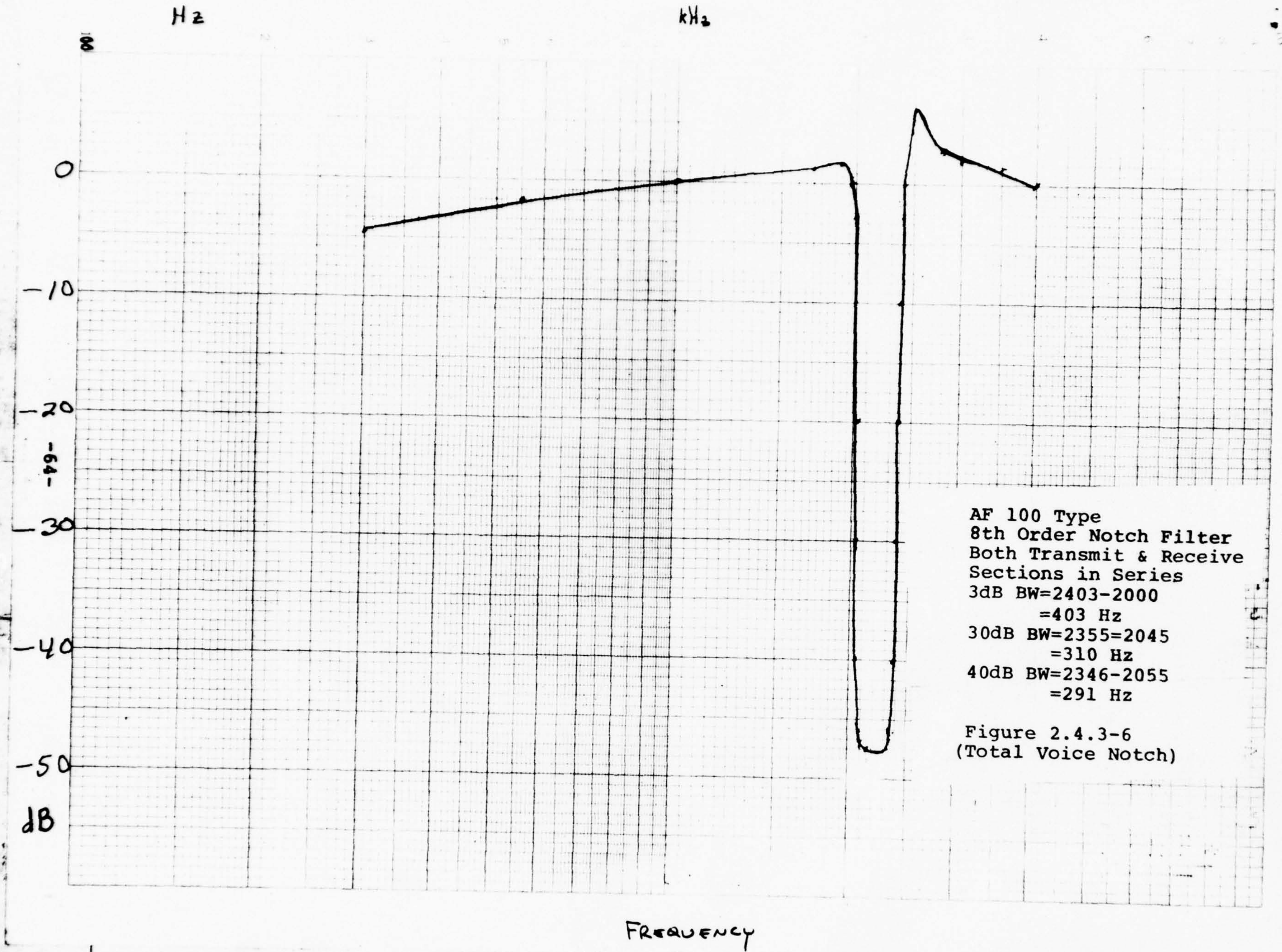
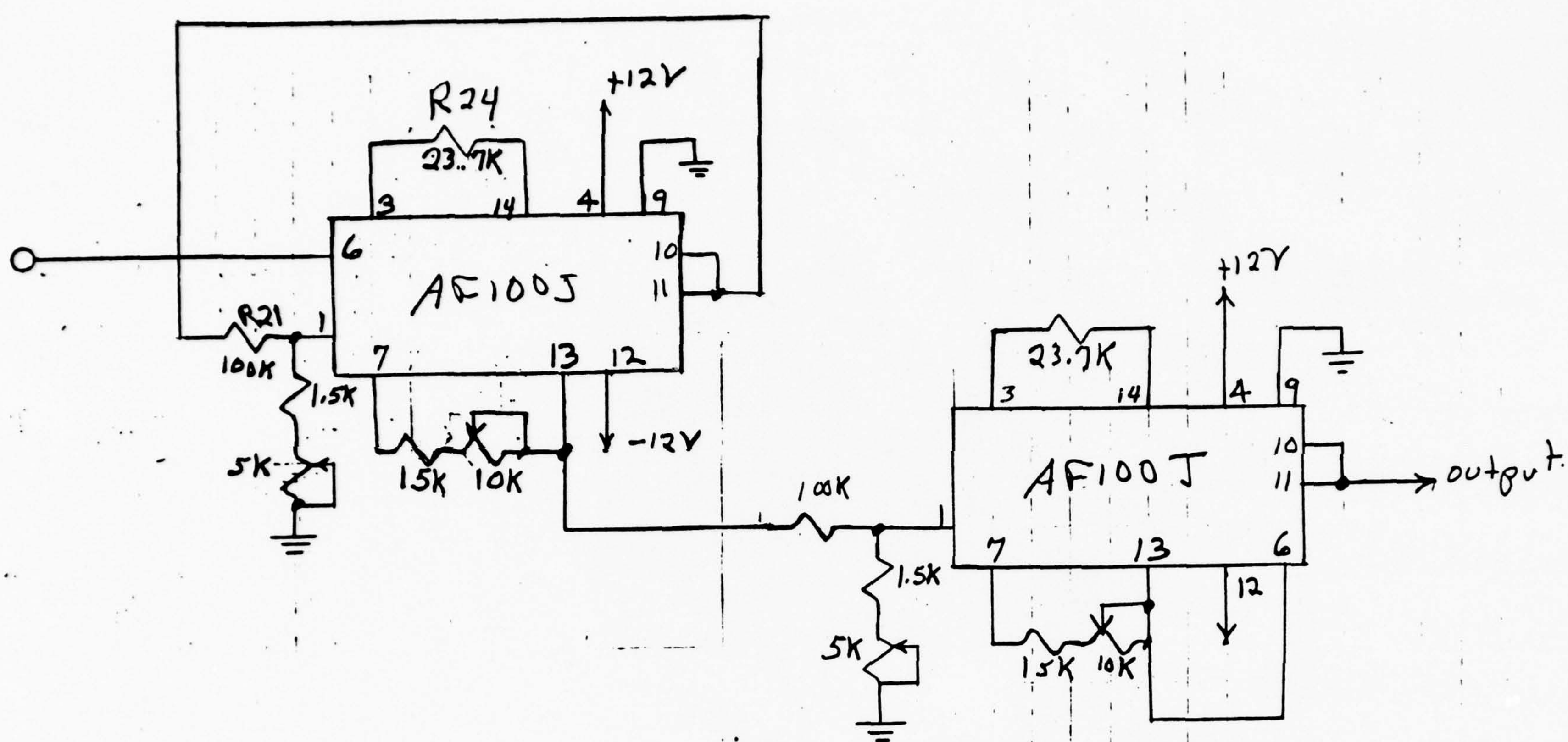


Figure 2.4.3-6
(Total Voice Notch)

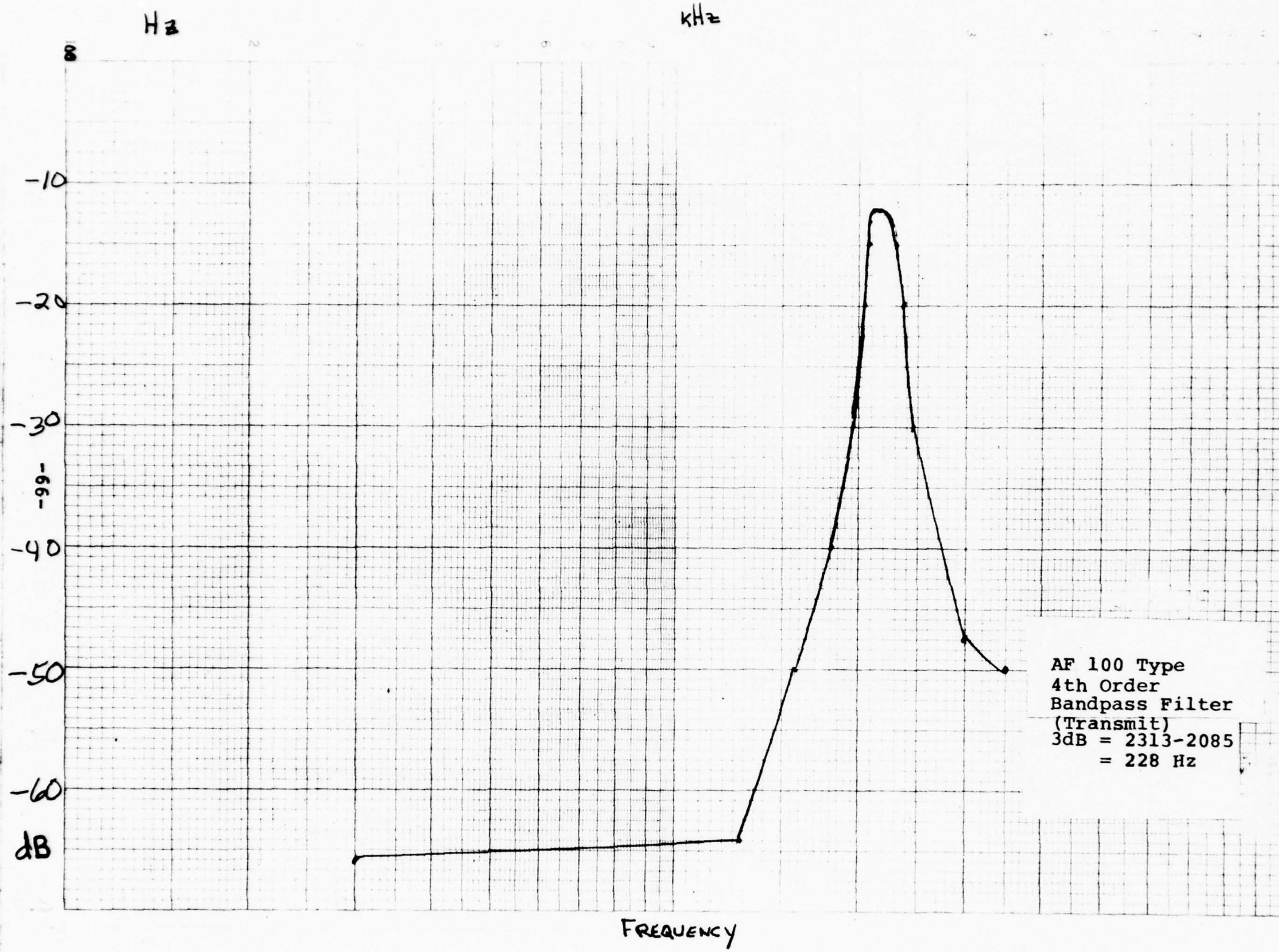


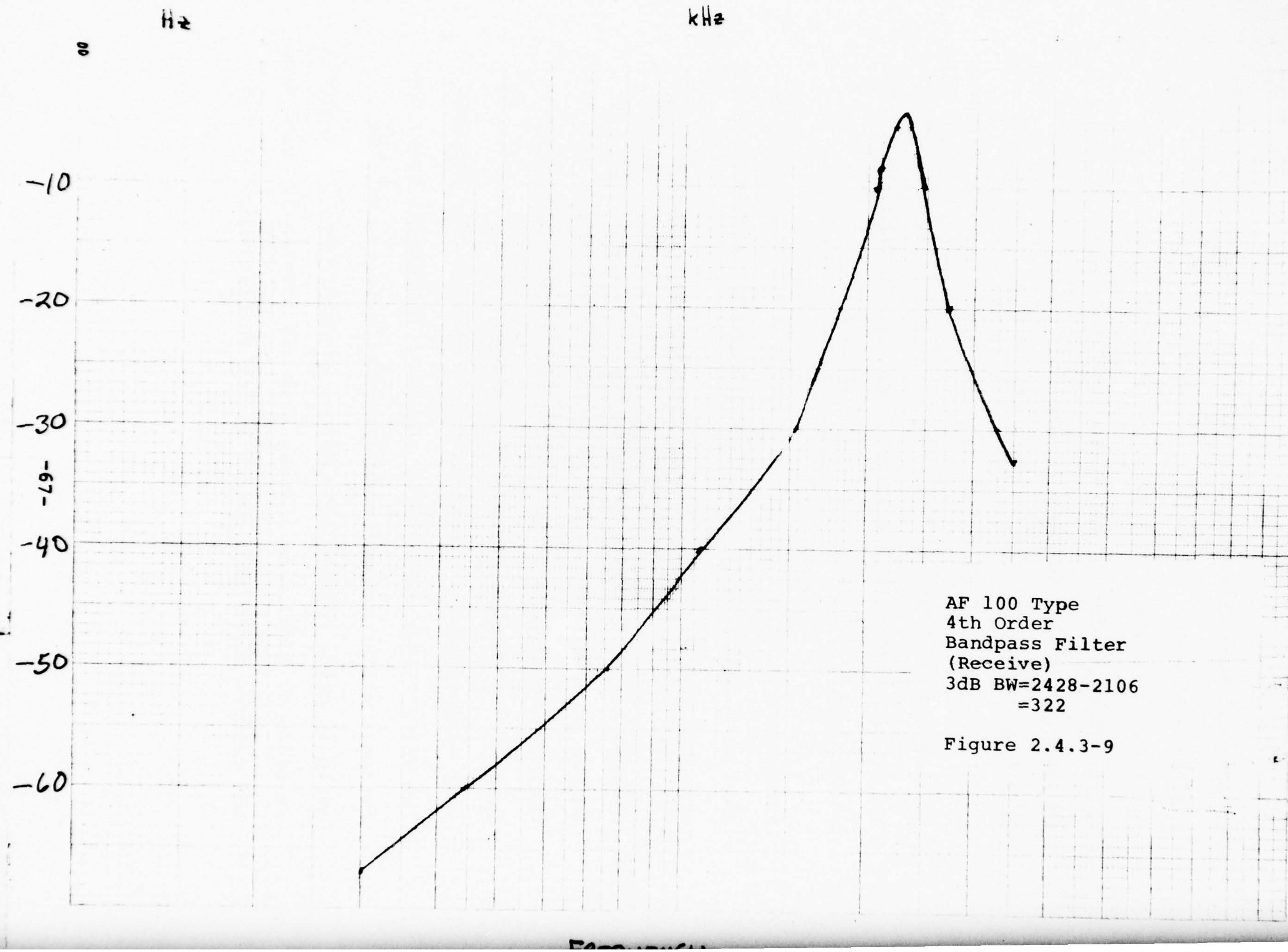
BANDPASS FILTER (ELLIPTICAL)

4th ORDER

(FOR TRANSMIT AND RECEIVE SECTIONS)

FIGURE 2.4.3 - 7


Hz
kHz
-10
-20
-30
-40
-50
-60
dB
FREQUENCY
AF 100 Type
4th Order
Bandpass Filter
(Transmit)
3dB = 2313-2085
= 228 Hz



AF 100 Type
4th Order
Bandpass Filter
(Receive)
3dB BW=2428-2106
=322

Figure 2.4.3-9

to transmit with coincidental bandwidth filters. Orion engineers expanded the concept at this point in lieu of concluding the study with negative results, and with the filter characteristics and system configuration that was implemented it was at least possible to collect meaningful data that would be useful to the FAA. (Note: More information on the transmission of data and voice,utilizing these filters,can be found in the results of the laboratory tests and in the theoretical aspects as written in paragraph 2.5 of this report.)

Furthermore, voice quality, although subjective, indicated good performance using both male and female speakers, even with the broader bandwidth notch.

2.4.4 SECTION 2.4 - SUMMARY & CONCLUSION

As can be determined by the results as indicated in the preceding paragraphs of this section, it is technically not feasible to provide acceptable voice plus data transmission when "coincidental" bandwidth filter networks are employed. The voice filter must employ a broader bandwidth, i.e., the bandpass filter network must essentially "fit into" the notch in order to eliminate mutual interference. Additionally, in order to provide data rates that can provide 80 mesec send and confirm times as required by the specification of this program, the bandpass filter must be made broad enough to accept the modulation products, which as stated in the "Modulation Technique" paragraphs are a function of the switching rate.

The filters designed in iteration three represent a near practical limit in terms of filters implemented with a universal active filter element. Passive networks can, of course, be considered; however, in order to maximize the packaging density of any future system implementation, an active filter is most desirable. In order to obtain more ideal responses, higher order filter designs will be required. To obtain these filters a custom design will be required. It is worthwhile at this point to perform additional analysis in order to gain more insight into ultimate filter requirements and practical attainability.

2.5 DATA RATE

In order to determine the maximum ideal data rate in a 240 Hz slot centered at 2210 Hz, an analysis was performed of PSK (bi-phase) modulation for the data signal at 150 bits per second and 200 bits per second data rates. Investigations were made into intersymbol interference in the data channel, power budget between voice and data signals, cross-talk between voice and data channels and error performance of the data channel. In the calculations, voice and data receive power levels were adjusted so that error performance of the data channel would be met and then the resulting data cross-talk into the voice channel calculated.

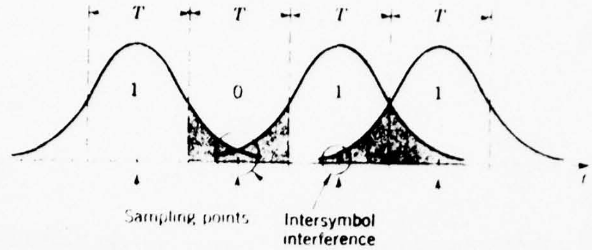
A 200 bit per second data rate is a reasonable goal for the PSK bi-phase modulation. The following analysis gives insight into channel capacity and ultimate filter requirements. The analysis is performed assuming filters with characteristics developed in iteration 3, as described in section 2.4.

"THE BALANCE OF THIS SHEET INTENTIONALLY LEFT BLANK"

2.5.1 INTERSYMBOL INTERFERENCE IN DATA CHANNEL

Signal overlaps into adjacent time slots may, if too strong, result in erroneous demodulation of the received data signal. This is termed intersymbol interference. Signal overlap is the result of spreading of the PSK pulses by the transmit and receive bandpass filters. Figure 2.5.1-1 demonstrates the effect of pulse spreading. The worst case is shown during the second phase where interference from two adjacent pulses adds.

The sharper the roll-off of the bandpass filters, the worse the spreading. Spreading can also be caused by nonlinear phase response of the bandpass filters themselves, or the telephone line, or line amplifiers in between. However, as seen from Figure 2.5.1-2, the phase response of the filters in the bandpass region is almost linear. Also, the phase characteristic of the telephone line in the same region is known to be linear. Therefore, the spreading due to filter roll-off is the predominant contribution to intersymbol interference.



Intersymbol interference in digital transmission.

FIGURE 2.5.1-1

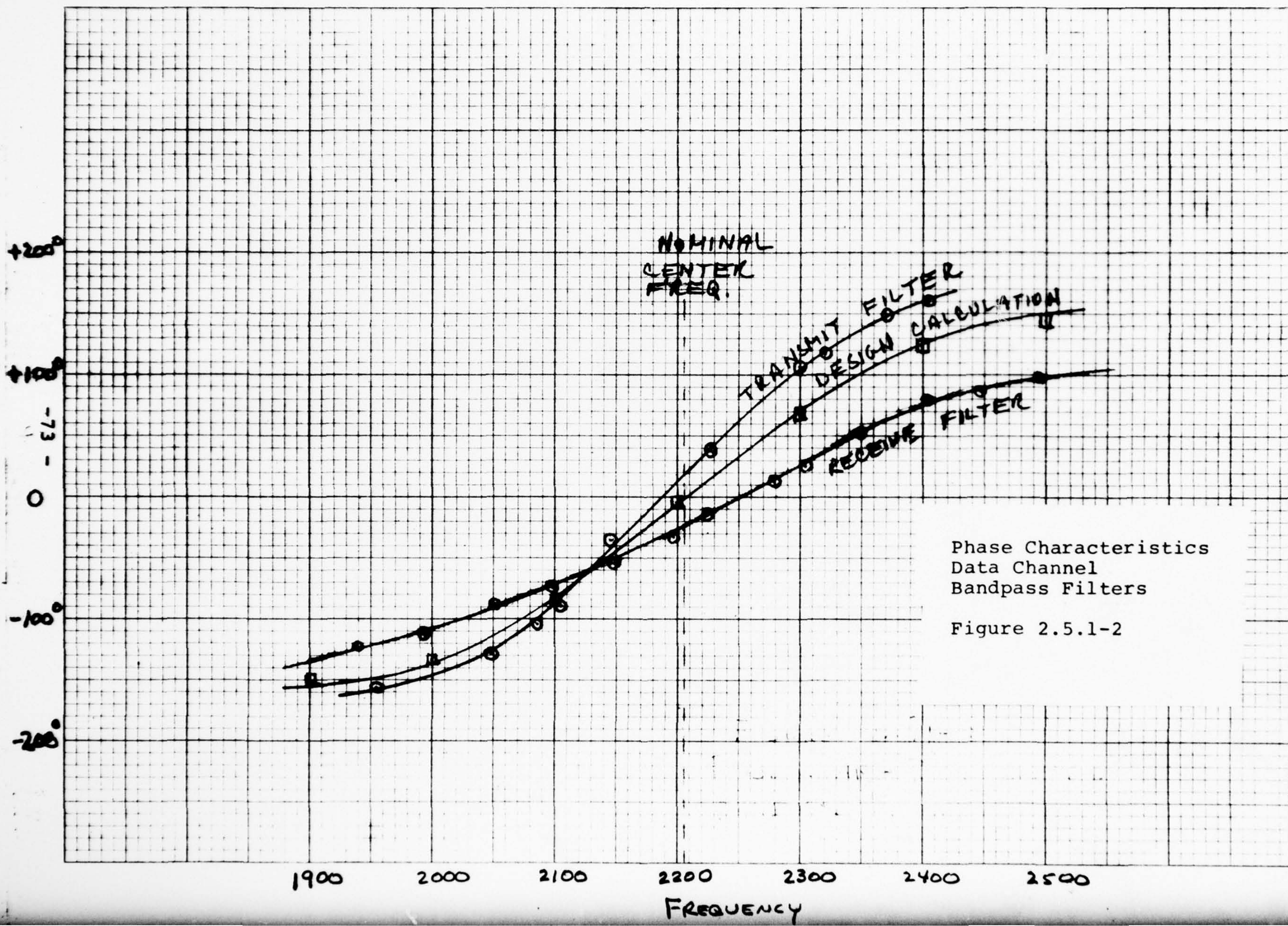
Determination of pulse spreading requires that the filter response to a rectangular PSK pulse be calculated. As this is difficult and time consuming, a worst case approximation will be used. That is, the filter pair will be assumed to have perfect cutoff (infinite roll-off) resulting in the worst possible spreading. The analysis for this worst case situation is given in a number of texts including Schwartz, "Information Transmission, Modulation and Noise" on page 61. (The analysis in Schwartz is for a rectangular d.c. pulse through a low pass filter, but the results for a rectangular PSK pulse centered in a bandpass filter would be the same.)

The envelope of each received PSK pulse after ideal filtering is given by:

$$g(t) = \frac{1}{\pi}\left\{Si\left[2\pi B\left(t+\frac{\tau}{2}\right)\right] - Si\left[2\pi B\left(t-\frac{\tau}{2}\right)\right]\right\}$$

where g(t) is normalized for a unity value pulse.

B is half the filter bandwidth (we will assume that this is at the 5dB points of either of the filters, which is 10dB down after both filters) = 150 Hz. t is the value of time relative to the center of the pulse, and τ is the pulse width for



Phase Characteristics
Data Channel
Bandpass Filters

Figure 2.5.1-2

which we will chose two trial values

$$\tau = \frac{1}{150} \text{ (case a) or } \frac{1}{200} \text{ (case b)}$$

for a data rate of 150 bps or 200 bps, respectively.

The interfering signal value from an adjacent pulse g(t) is at t = τ (one pulse width away). Thus, we are interested in

$$g(\tau) = \frac{1}{\pi} Si[2\pi B\tau(3/2)] - Si[2\pi B\tau(1/2)]$$

for case a. $B\tau = 1$

case b. $B\tau = 3/4$

case a.

$$g(\tau) = \frac{1}{\pi}[Si(3\pi) - Si(\pi)] = \frac{1.67-1.85}{3.14} = -.057$$

case b.

$$g(\tau) = \frac{1}{\pi}[Si(2.25\pi) - Si(.75\pi)] = \frac{1.45-1.74}{3.14}$$

$$= -.092$$

Now, since the worst case occurs when the interference form two adjacent pulses (one on either side) contribute to moving the signal level, the interfering signal values must be doubled. That is,the maximum interference is -.114 or -.184 for case a and b, respectively. Thus, effective signal power is reduced by 1.1dB for case a and 1.8dB for case b. A margin of these amounts will have to be added to the data signal power to achieve the desired error performance.

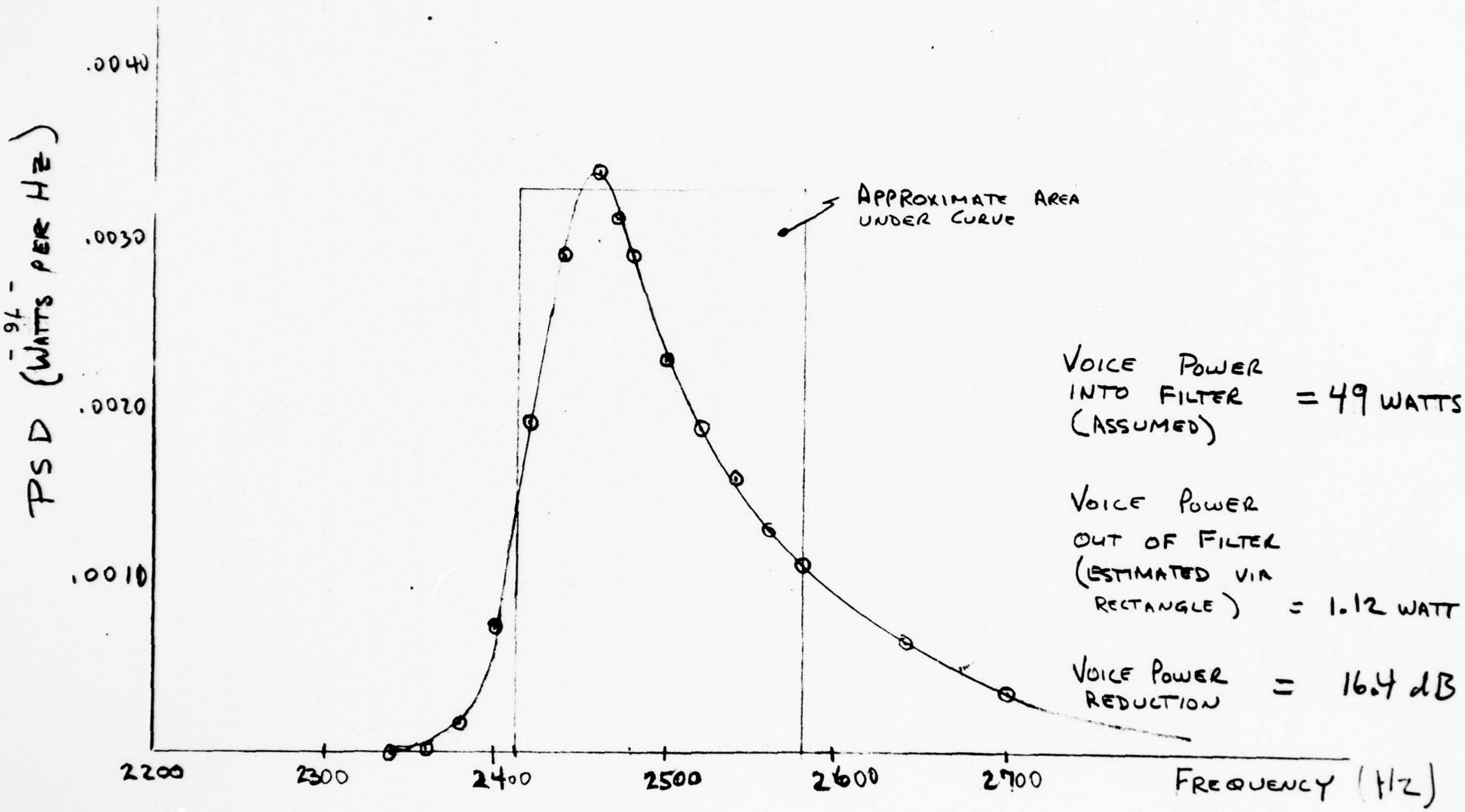
The interfering signals act to reduce the signal strength of the desired signals when and only when three adjacent pulses are all of the same phase (i.e., the same bit pattern.) Though this event occurs only 25% of the time, for a random data stream, the margin required to overcome effective loss in signal power is not much less than the 1.1dB and 1.8dB values calculated above.

2.5.2 CROSS TALK OF VOICE INTO THE DATA CHANNEL

Figure 2.5.2-1 is a plot of the power spectral density of voice signal that leaks through the data channel receive bandpass filter and therefore enters the PSK detector. It was calculated using the actual filter characteristics of the experimental equipment. The plotted values are based on an assumed 49 watts voice power input to the bandpass (BP) filter, but this is easily scaled to any other input power assumption. The area under the curve represents the power of the voice cross-talk into the data channel PSK detector. It turns out that the cross-talk power is much greater than the line noise in the BP filter even when power levels are adjusted to actual values. The important result of this calculation is the ratio of the voice signal power leaking through the BP filter to the voice power into the filter. This ratio is 16.4dB.

Figure 2.5.2-1

Voice Signal Power Spectral Density out of
Receiver Data Bandpass Filter



2.5.3 ERROR PERFORMANCE AND POWER BUDGET

To achieve a one in 10^6 bit error rate an $\frac{E_b}{N_O}$ (signal-energy-per-bit to noise-power-density) ratio of approximately 11.4dB is required* assuming the telephone channel is disturbed only by Gaussian noise.

In order to determine the relative power that must be budgeted for the data signal required to achieve this $\frac{E_b}{N_O}$, the data signal power ratio to noise power ($\frac{Sd}{N}$) in the data channel (300 Hz) bandwidth must be calculated. This turns out to be 8.4dB for a 150 bps data rate and 9.6dB for a 200 bps data rate. Adding the margins required to overcome intersymbol interference, the data signal-to-noise values over the channel bandwidth become 9.5dB and 11.4dB, respectively.

Since the predominant interference in the data channel is the cross-talk of the voice signal into the 300 Hz data channel receive filter, additional margin will have to be provided. This is necessary because of the bursty nature of the voice signal. An additional 6dB margin should suffice. Thus, for a 150 bps data rate the $\frac{Sd}{N}$ required is 15.5 dB where N is the average power in the voice cross-talk getting through the data channel receive filter. For a 200 bps data rate the $\frac{Sd}{N}$ required is 17.4 dB.

*This is found from an extrapolation of the error rate curve for differentially coherent PSK in Swanson, "Phase Shift Keying," in the July 1962 issue of Space/Aeronautics.

In section 2.5.2, it was determined that the cross-talk leakage of the voice signal into the data channel receive filter is 16.4 dB down from the received voice signal power Sv. Therefore, the voice signal power in dB must be adjusted so that:

$$Sv = Sd \ -15.5 \text{ dB} + 16.4 \text{ dB for 150 bps}$$
$$= Sd \ -17.4 \text{ dB} + 16.4 \text{ dB for 200 bps}$$

Thus, voice signal power before the receive filters should be adjusted to about 1 dB higher than the data signal power at the same points for the 150 bps data rate and 1 dB less than the data signal power for the 200 bps rate.

2.5.4 CROSS TALK OF DATA SIGNAL INTO THE VOICE CHANNEL

a. Spectral Properties of a Random Pulse Train

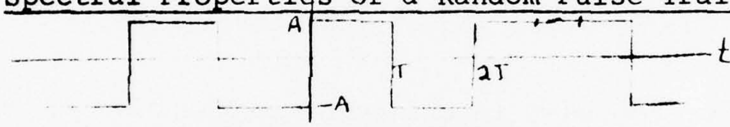


Figure 2.5.4-1 Binary Data Signal

The autocorrelation function is given by: $R_x(\tau) = A^2\left(1 - \frac{|\tau|}{T}\right)$ for $|\tau| < T$

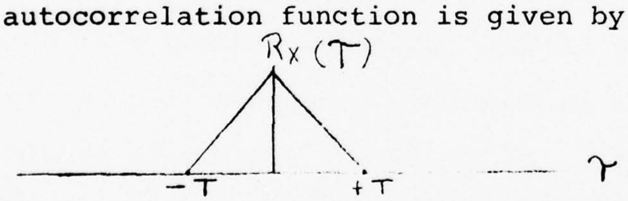


Figure 2.5.4-2 Autocorrelation of Binary Data Signal

The power density spectrum S_x (w) is then

$$S_x(w) = \mathfrak{F}[R_x(\tau)] = TA^2 Sa^2\left(\frac{wT}{2}\right)$$

Figure 2.5.4-3 Power Density Spectrum (PSD) of Binary Data Signal

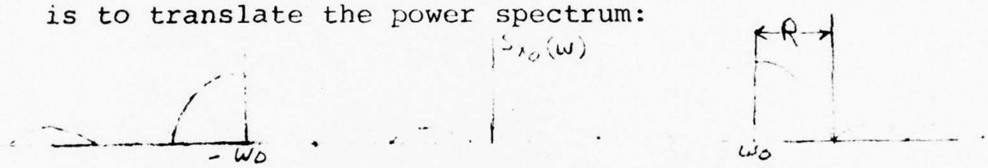
b. <u>Bandwidth vs. Signalling Rate</u>

For a signalling rate of R bits per second, the pulse width, T, is $T = \frac{1}{R}$

Hence, if R b/s are transmitted, the first zero-crossing of the spectrum is at frequency f=R.

c. <u>Modulation</u>

If the random pulse train in section 2.5.4.a modulates a continuous wave (CW), the resultant modulated signal is phase shift keying (PSK). The effect of the amplitude modulation is to translate the power spectrum:



<u>Figure 2.5.4-4 - PSD of Modulated PSK Data Signal</u>

Thus, the power density spectrum of the modulated signal is symmetric about the carrier frequency.

d. <u>Bandpass Filtering</u>

As a result of bandpass filtering at the transmitter, the power density spectrum of the modulated data signal is modified. The resulting power density spectrums of filtered data signals, for the cases R=150 b/s and R=200 b/s, is given in Tables I and II, respectively. A plot of the filtered data-signal power density spectrum after BP filtering is shown in Figure 2.5.4-5.

TABLE I - PSD of 150 bps PSK Data Signal Before and After Filtering

Signal Spectrum		After Transmit BP Filter		After Receive BR Filter	
f	Power	f	Power	f	Power
2210	0 dB	2210	-13.0dB	2210	< -50 dB
2220	-.06	2210	-13.06	2220	"
2230	-.26	2230	-13.26	2230	"
2240	-.58	2240	-13.58	2240	"
2250	-1.04	2250	-14.04	2250	"
2260	-1.65	2260	-14.65	2260	"
2270	-2.42	2270	-15.42	2270	"
2280	-3.37	2280	-16.37	2280	"
2290	-4.53	2290	-17.53	2290	"
2300	-5.93	2300	-19.13	2300	"
2310	-7.67	2310	-22.17	2310	"
2320	-9.83	2320	-25.33	2320	"
2330	-12.62	2330	-29.12	2330	"
2340	-16.51	2340	-34.51	2340	"
2350	-22.99	2350	-41.99	2350	"
2360	$-\infty$	2360	$-\infty$	2360	"
2370	-24.15	2370	-45.15	2370	"
2380	-18.84	2380	-40.84	2380	< -50 dB
2390	-16.14	2390	-39.14	2390	-47.14
2400	-14.57	2400	-38.57	2400	-43.50
2410	-13.69	2410	-38.69	2410	-40.69
2420	-13.30	2420	-39.3	2420	-39.3
2430	-13.32	2430	-40.32	2430	-39.3

TABLE I (Continued)

Signal Spectrum		After Transmit BP Filter		After Receive BR Filter	
f	Power	f	Power	f	Power
2440	-13.70	2440	-41.5	2440	-39.5
2450	-14.46	2450	-42.96	2450	-40.46
2460	—	—	—	—	
2470	-17.30	2470	-47.30	2470	-44.80
2480	—	—	—	—	
2490	-23.18	2490	-54.18	2490	-51.68
2500	—	—	—	—	—
2510	$-\infty$	2510	$-\infty$	2510	$-\infty$

TABLE II - PSD of 200 bps PSK Data Signal Before and After Filtering

Signal Spectrum		After Transmit BP Filter		After Receive BR Filter	
f	Power	f	Power	f	Power
2210	0 dB	2210	-13	2210	<-60 dB
2220	-.04	2220	-13.04	2220	"
2230	-.14	2230		2230	"
2240	-.32	2240	-13.32	2240	"
2250	-.58	2250		2250	"
2260	-.91	2260	-14.00	2260	"
2270	-1.33	2270		2270	"
2280	-1.83	2280	-15.00	2280	"
2290	-2.42	2290		2290	"
2300	-3.11	2300		2300	"
2310	-3.92	2310	-18.92	2310	"
2320	-4.86	2320	-20.36	2320	<-60.36
2330	-5.94	2330	-22.44	2330	-55.44
2340	-7.20	2340	-24.20	2340	-52.20
2350	-8.69	2350	-27.69	2350	-50.19
2360	-10.45	2360	-30.45	2360	-48.45
2370	-12.62	2370	-33.62	2370	-48.62
2380	-15.39	2380	-37.39	2380	-49.39
2390	-19.23	2390	-42.23	2390	50.23
2400	-25.61	2400	-49.61	2400	57.61
2410	$-\infty$	2410	$-\infty$	2410	$-\infty$
2420	-26.48	2420	-52.48	2420	-52.48
2430	-20.97	2430	-47.97	2430	-49.47
2440	-18.02	2440	-46.02	2440	-44.02

TABLE II (Continued)

Signal Spectrum		After Transmit BP Filter		After Receive BR Filter	
f	Power	f	Power	f	Power
2450	-16.14	2450	-44.64	2450	-42.14
2460	-10.45	2460	-39.70	2460	-37.20
2470	-14.06	2470	-44.06	2470	-41.56
2480	-15.39	2480	-46.89	2480	-44.39
2490	-19.23	2490	-50.23	2490	-47.73
2500	-25.61	2500	-57.11	2500	-55.11
2510	$-\infty$	2510	$-\infty$	2510	$-\infty$

e. Band Reject Filtering

The receiver band reject (BR) filter attempts to separate the data signal from the voice signal. Unfortunately, this is not entirely possible. To determine just how much of the data signal is present in the voice band after BR filtering, the following analysis is given.

As shown in section 2.5.3, for the 150 bps data rate, if all of the data signal got into the voice band, the ratio of data signal power-to-voice signal power would be -1dB. This, of course, would be unacceptable. Therefore, the effects of the BR filter must be analyzed.

Denote the power of the data signal at the input to the BR filter as S_D, and the power of the same signal after BR filter as S_{DR}. The power S_{DR} remains in the voice band. In figure 2.5.4-6A, the power density spectrum of the input data signal is plotted for the case R=150 b/s. The area under this curve is the signal power S_D. The power density spectrum of the signal at the output of BR filter is sketched in Figure 2.5.4-6B. The area under this curve is S_{DR}. The ratio of these two powers is -29.40, which indicates how much of the data signal has been rejected. Since the ratio of S_D to the received voice signal power is -1dB, the remaining data signal after BR filtering is -29.4 - 1 = -30.4 dB.

The same analysis is summarized in Figures 2.5.4-7A and

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STUDY PROGRAM FOR ANALYZING DATA HANDLING CAPABILITY OF A 240 H--ETC(U)
AUG 78 F AFFELDT, R MOORS, C KULESZA DOT-FA78WA-4099
UNCLASSIFIED OSI-78-FAA-SS FAA/RD-78/116 NL

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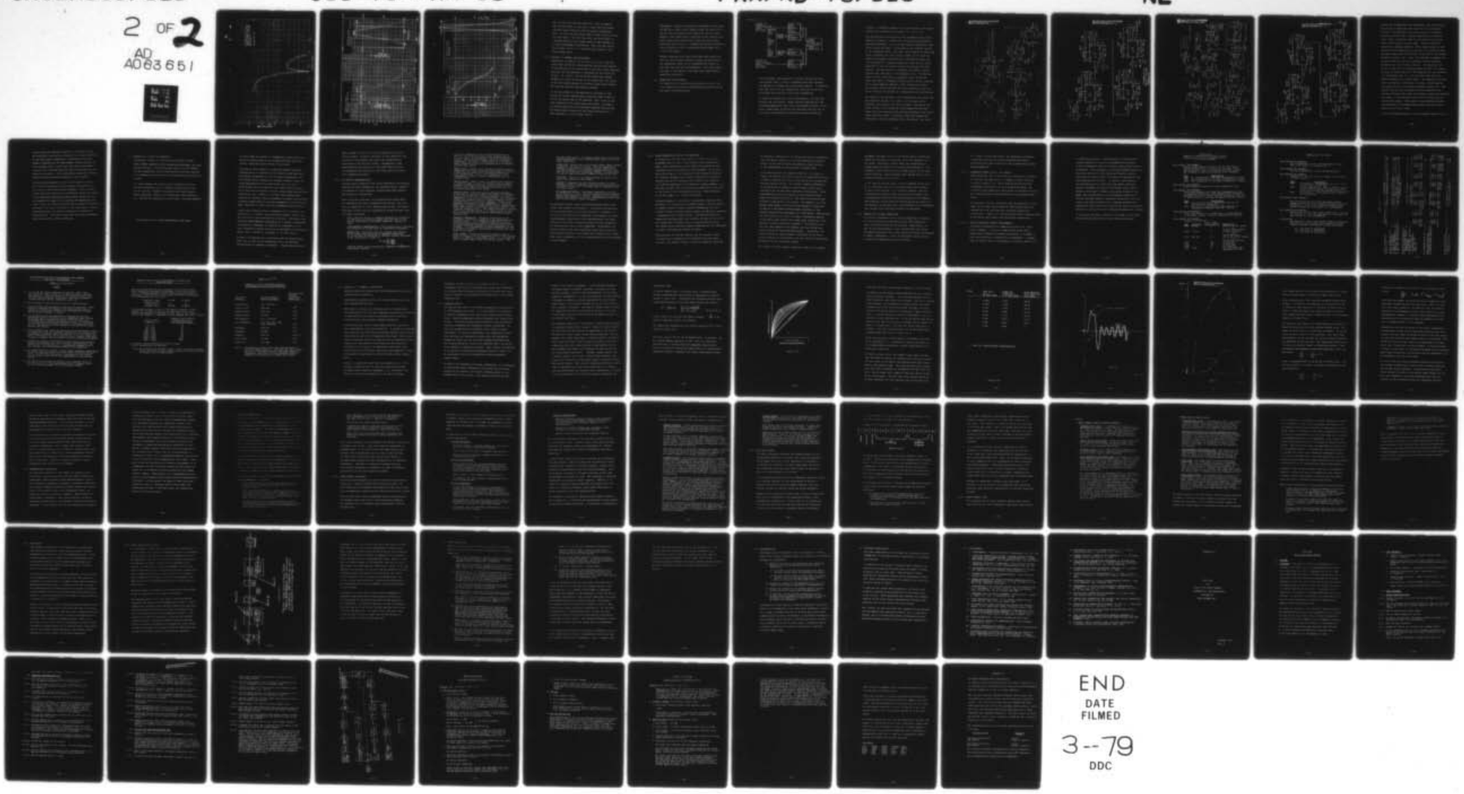

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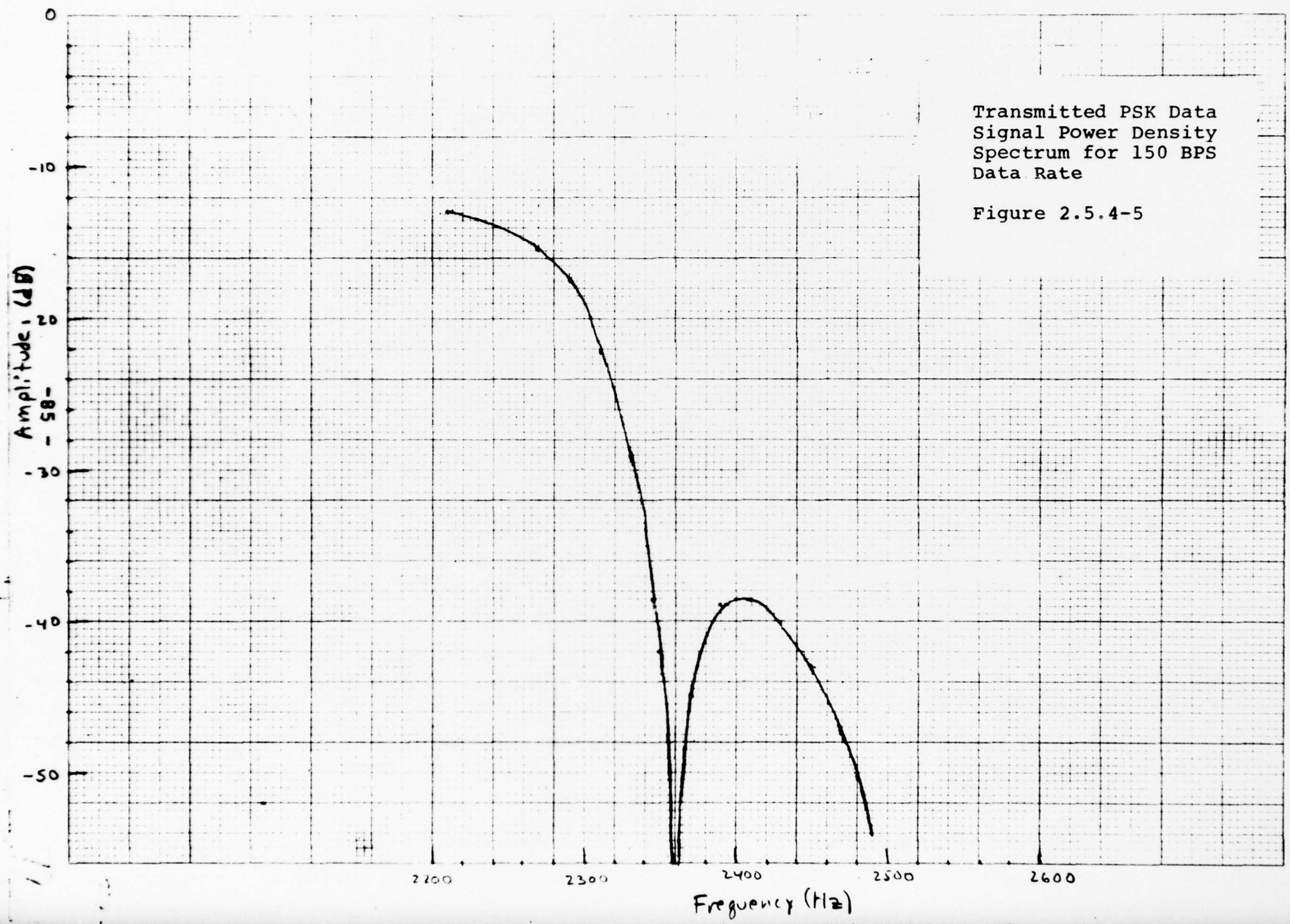


Transmitted PSK Data Signal Power Density Spectrum for 150 BPS Data Rate

Figure 2.5.4-5

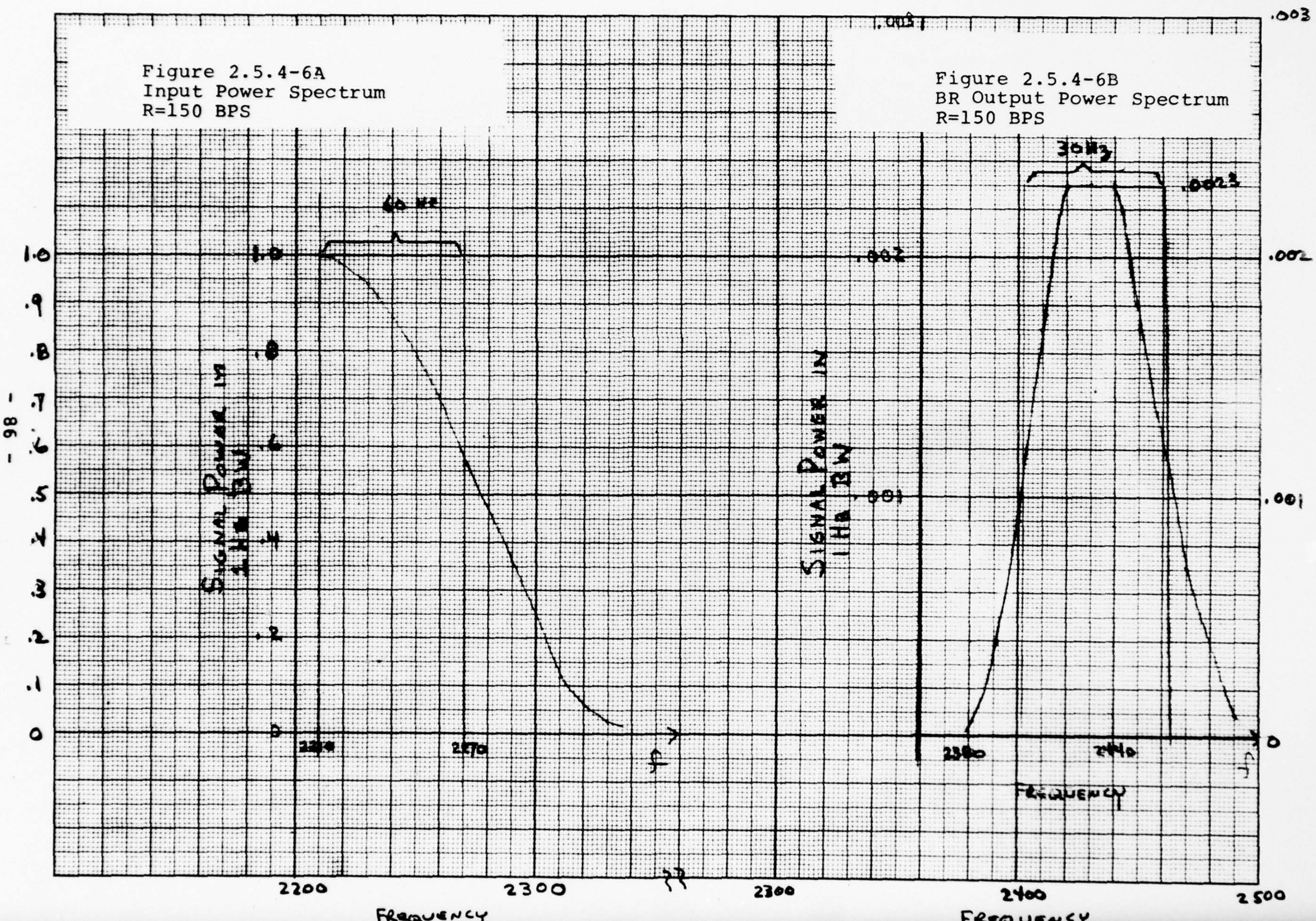


Figure 2.5.4-6A
Input Power Spectrum
R=150 BPS

Figure 2.5.4-6B
BR Output Power Spectrum
R=150 BPS

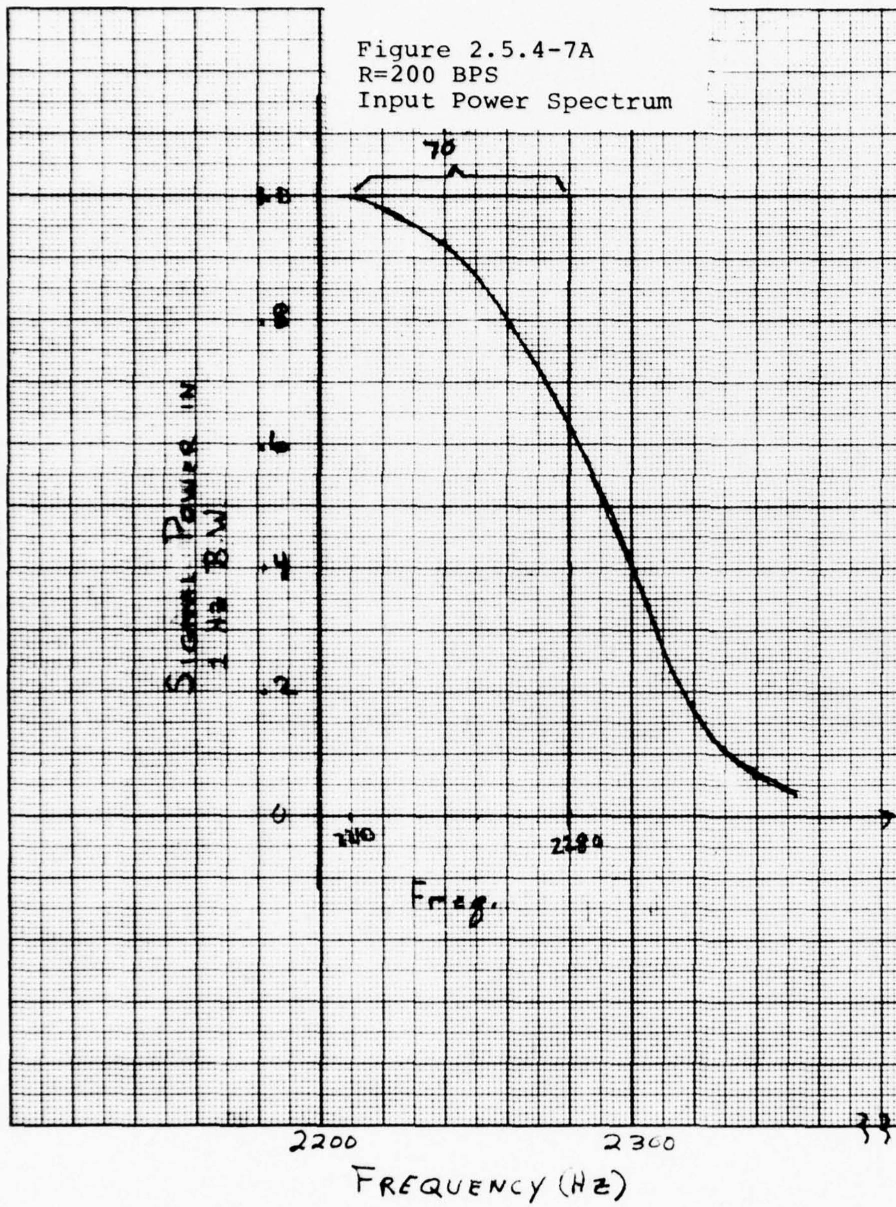


Figure 2.5.4-7A
R=200 BPS
Input Power Spectrum

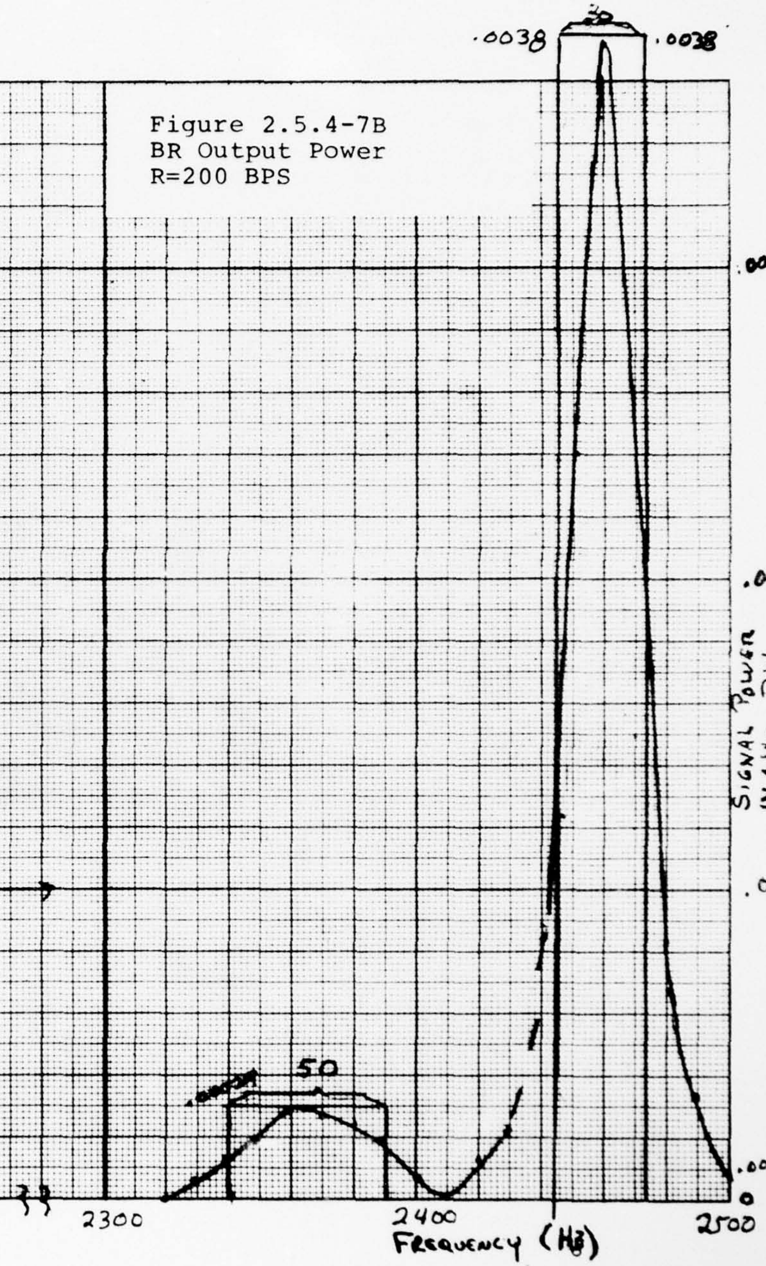


Figure 2.5.4-7B
BR Output Power
R=200 BPS

and 2.5.4-7B, for the case R=200 b/s. Here we observe that the BR filter is less able to cope with the broader spectrum signal resulting from the higher transmission rate. In this case the output power is down -27.36 dB from the input power. For this data rate, the data signal is 1 dB more than the voice signal power to meet probability of error performance requirements. Thus, the data signal will be down -27.36 + 1 = -26.36 dB with respect to the voice signal power.

2.5.5 Section 2.5 Summary and Conclusions

Examination of Figures 2.5.4-6 and 2.5.4-7 show that the data signal cross-talk remaining after receive BR filtering is due entirely to the second lobe of the data signal spectrum. Even part of this spectrum is attenuated by the receive BR filter. If, in fact, the receive BR filter were widened slightly to reject the center of the second lobe, the amount of the data signal present in the voice signal after receive BR filtering could be significantly reduced.

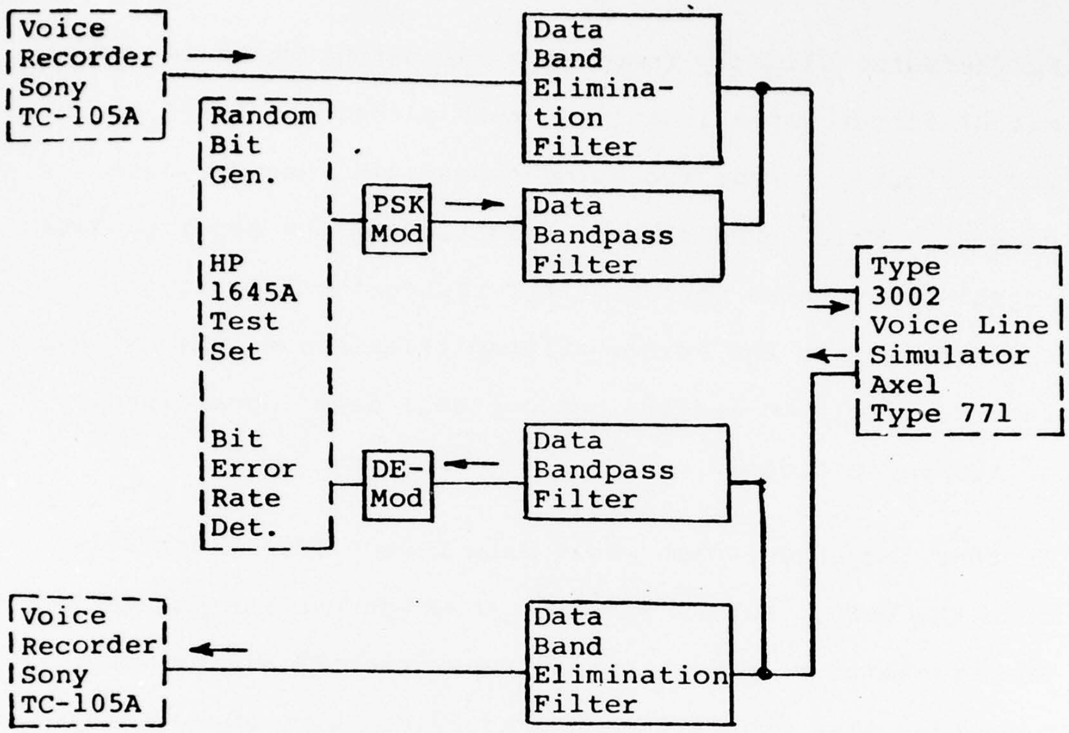
In the case R=200 b/s, the spectrum at the output of the receive BR filter contains part of the main lobe as well as all of the second lobe of the spectrum. This is due to the wider spectrum of the higher data rate signal. To reject the primary and most of the secondary spectral lobes at 200 bps would require widening of the receive BR filter to a half bandwidth at 3 dB of about 170 Hz.

Furthermore, slightly increasing the bandwidth of the transmit BR filter would greatly improve either the R=150 b/s or 200 b/s case by reducing voice cross-talk into the data channel. This would allow a reduction in the required data signal power which would further reduce the data signal cross-talk into the voice. Transmitting beyond the R=200 b/s rate is likely to lead to unacceptable data signal levels in the voice signal.

Another technique which would also reduce data cross-talk into the voice channel would be to shape the PSK pulses in the transmitter (prior to the transmitter BP filter) so that they rise and fall more gradually. This would reduce the power in the spectral side lobes which would further reduce the cross-talk.

2.6 SYSTEM TEST CONFIGURATION

The feasibility of transmitting data within a slot in the voice channel was tested by constructing and evaluating the test model illustrated below.



Voice recorders were employed to transmit and receive voice intelligence, while a Data-bit Generator/Error Rate Detector was employed to transmit and receive data signals. A simulated Type 3002 voice line was injected into the test set-up, over which both voice and data signals were transmitted.

To assess the capability of this configuration, circuits were designed and constructed. These circuits consisted of a PSK modulator/demodulator, band reject and bandpass filters, a line driver, amplifiers, flip/flops and interface circuits, all of which were interconnected to provide an overall system for testing. A schematic diagram of both the transmit and receive sections are illustrated in Figures 2.6-1 thru 2.6-4.

The circuits were constructed on four breadboard type circuit boards (1 transmit, 1 receive and 2 filters),interconnected and tests performed. As can be seen from the schematics, each signal (voice and data) is connected to the transmit section at two different inputs. The input voice signals are amplified (U1A), connected to a notch filter (Figure 2.6-2) where the signal is stripped of frequencies in the 2200 Hz range, and then connected to a summation amplifier (U1B) prior to being connected to the line driver circuits (Q1 and Q2) for transmission. The data signal is connected to a buffer inverter (U2A and Q3), and then to a PSK modulator (U3). The PSK modulator is then connected to a bandpass filter (U4 and U5) where the signal is bandwidth limited prior to being connected to the same summation amplifier (U1B) as the voice signal. Both signals (voice and data) are then connected to the line driver circuits (Q1 and Q2) for transmission over the wireline.

The output of the wireline is then connected to the receive input amplifier (Figure 2.6-3, U1A). At the output of the amplifier, both signals are fed to filters. One filter, the notch (Figure 2.6-4) strips the voice signal from the data signal with the resulting voice signals connected to the voice output amplifier (U1B). Conversely, when both signals are connected to the data bandpass filter (U3 and U4), the voice

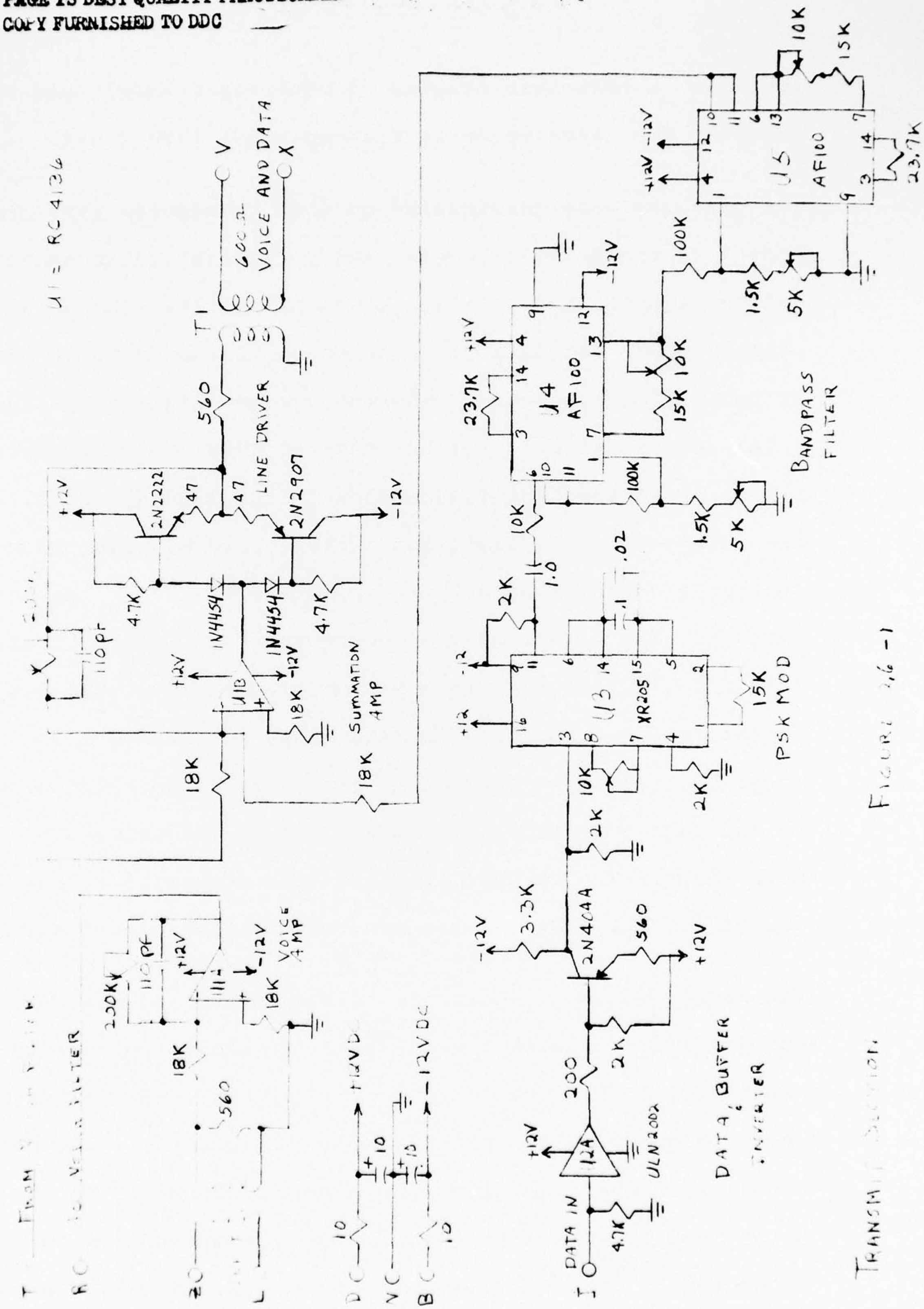


Figure 4.6-1

Transmit Section

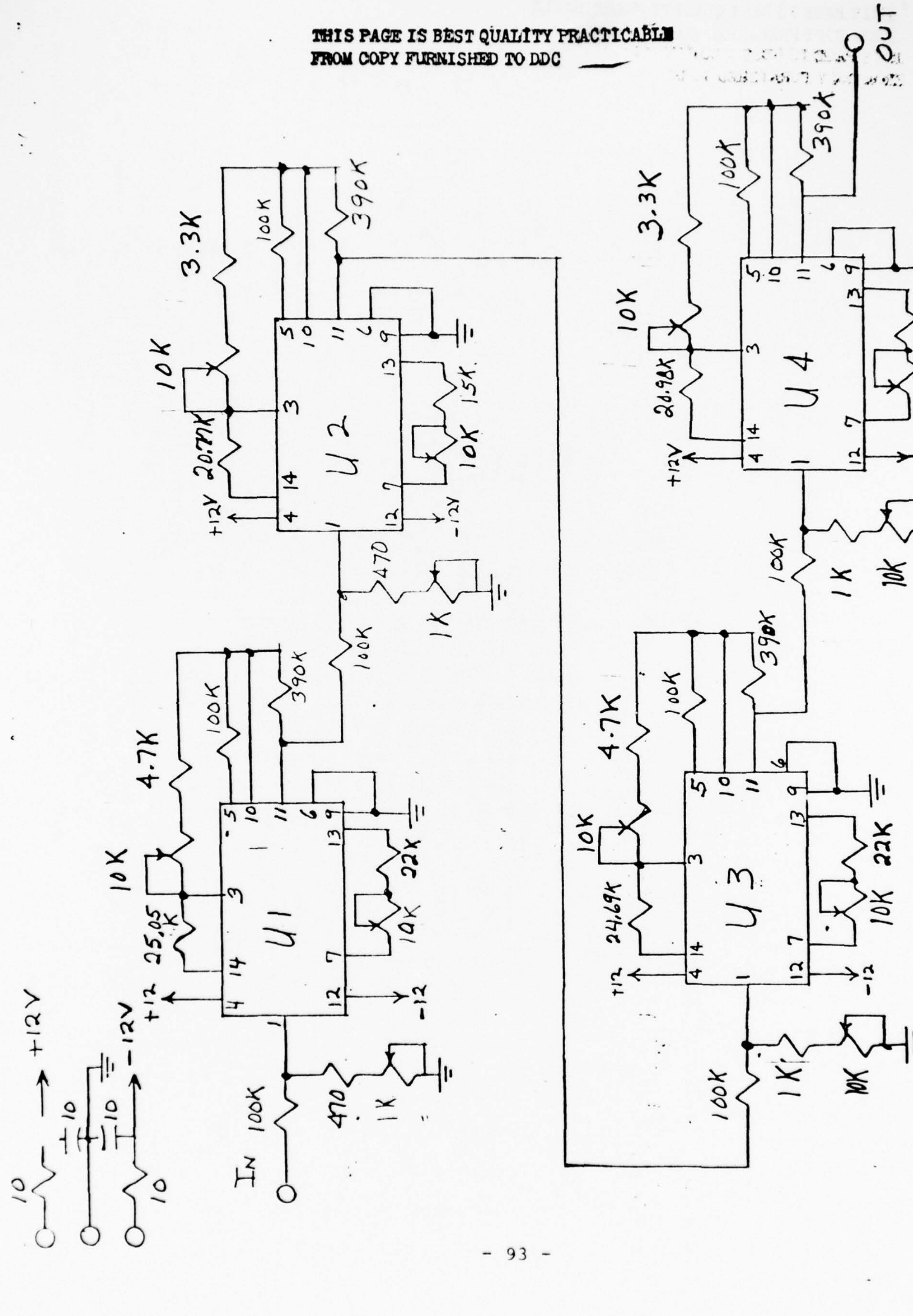


FIGURE 2.6-2
NOTCH FILTER
(TRANSMIT

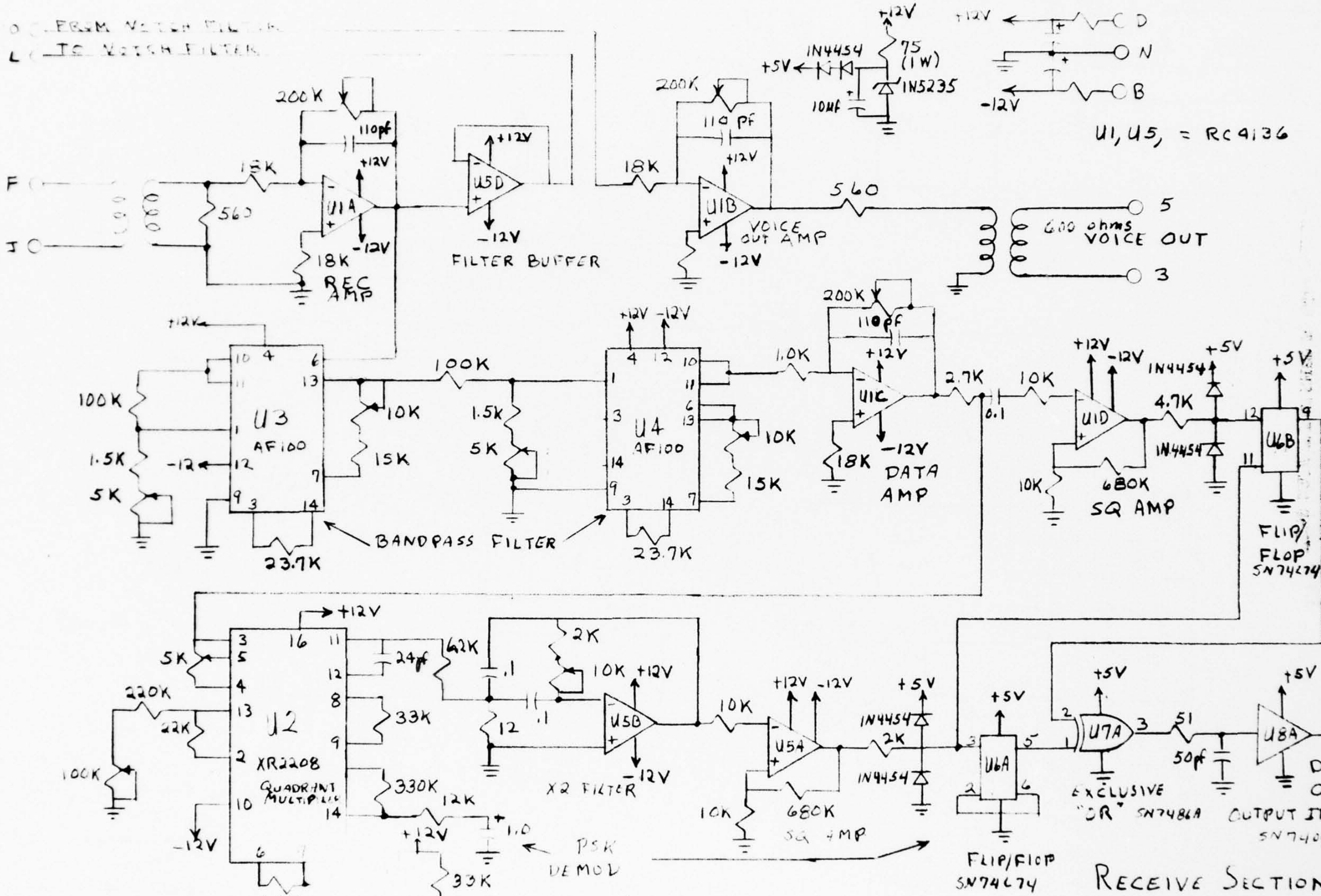

FROM NOTCH FILTER
TO NOTCH FILTER
REC AMP
FILTER BUFFER
VOICE OUT AMP
600 ohms VOICE OUT
U1, U5, = RC4136
U3 AF100
U4 AF100
BANDPASS FILTER
DATA AMP
SQ AMP
FLIP/FLOP SN74C74
U2 XR2208 QUADRANT MULTIPLIER
X2 FILTER
PSK DEMOD
FLIP/FLOP SN74C74
EXCLUSIVE "OR" SN7486A
OUTPUT INV SN7404
DATA OUT
RECEIVE SECTION

FIGURE 26-3

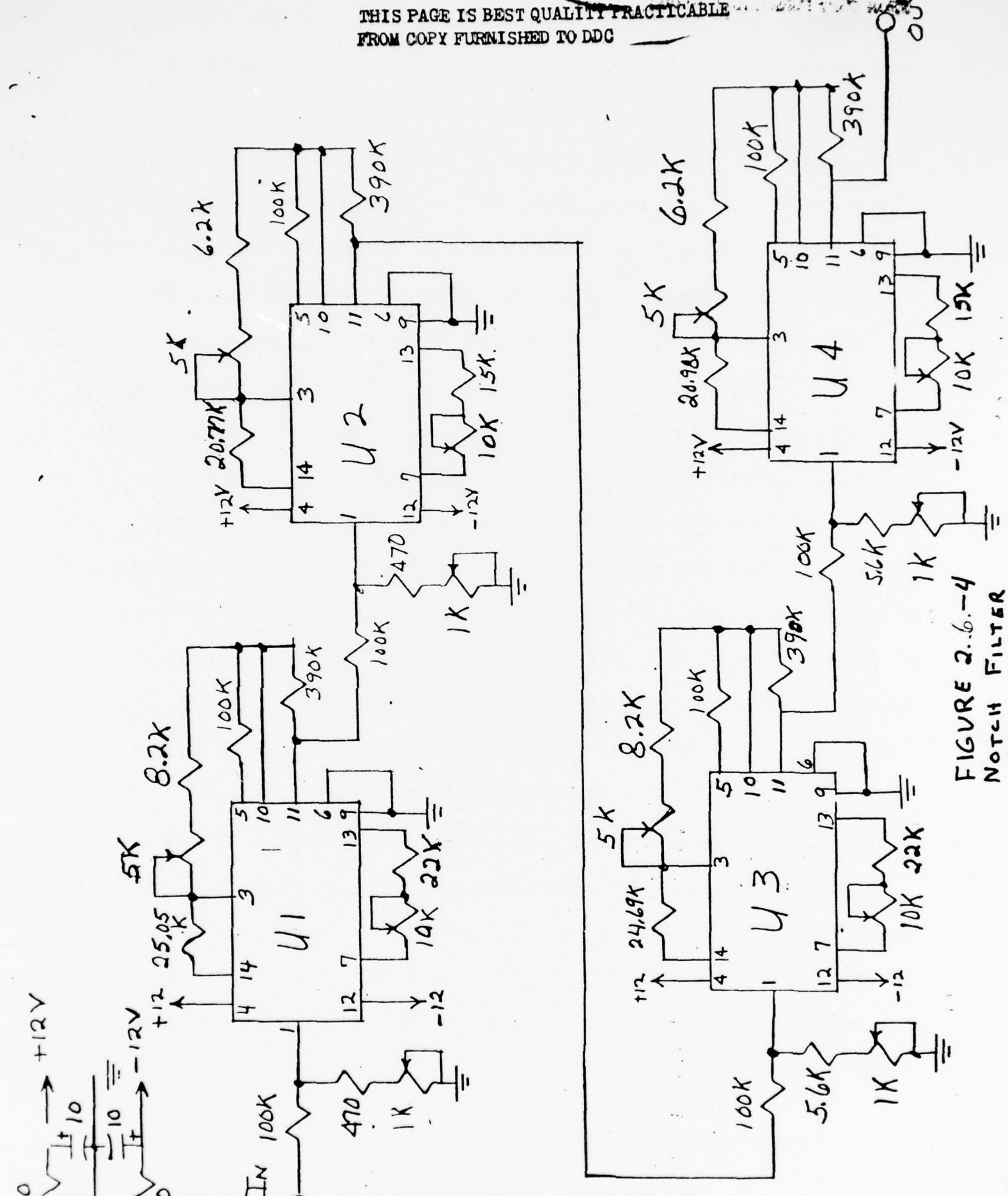


FIGURE 2.6.-4
NOTCH FILTER
(RECEIVE)

signals are stripped from the data signal, and the resulting data signal is amplified (U1C) and fed to the demodulator circuits. The demodulator circuits consist of a quadrant multiplier circuit (U2), squaring amplifiers (U1D and U5A), filter (U5B), flip/flops (U6A, B), an exclusive OR circuit (U7A), and an output inverter (U8A). The data signal is squared by a squaring amplifier (U1D), whose output is diode limited for TTL compatibility, and then to a flip/flop (U6B). The same signal that is fed to U1D is also fed to the quadrant multiplier circuit (U2). Here, the frequency is doubled and filtered (U5B) and used as a reference carrier (coherent detection) for the recovery circuits. The output of the filter is then fed to a squaring amplifier (U5A), the output of which is diode limited for TTL compatibility and connected to 2 flip/flops (U6A, B). One flip/flop (U6B) uses this doubled frequency as a clock signal. The remaining flip/flop (U6A) is basically a divide by 2 circuits, thereby placing the signal back to its original frequency or data rate. The outputs from the two flip/flops are then compared by the exclusive OR gate (U7A) where the changes in phase are detected. The output of the exclusive OR then is in reality the recovered data signal. This signal is then buffered/inverted and represents the recovered data output which was used to detect the bit error rates.

As can be determined from the circuit diagram (Figure 2.6-3), inverted data was sometimes observed at the output because the flip/flops state were not always set to a reset condition at the end of each transmission. This condition did not present a problem with the experimental circuits because the test equipment used to detect and count error rates merely indicated this inverted condition, but did not count the data received as an error. In an operational system this inverted condition could be resolved simply by a reset signal.

The modulator/demodulator circuits are designed to operate with the HP 1645 Digital Pattern Analyzer and therefore do not reflect a final system design approach. This is because certain necessary functions are provided by the HP 1645 itself. A final modulator/demodulator design should include circuits to integrate the output of the existing PSK demodulator over about a fifth of a data pulse-width and to sample the resulting waveform at (or near) the center of each data pulse. A narrow band phase-locked loop would be required for clocking the sampling pulse. This would assure that the actual performance would be as close to ideal as possible.

2.7 TRANSMISSION SYSTEM CONSIDERATIONS

Transmission systems are basically derived from two media types; namely, physical, which include wireline/cable, and radio derived circuits. In most long haul cases it is not uncommon to have transmission circuits derived from facilities composed of a combination of both physical and radio derived transmission systems.

In a radio system, the effects on the reliability of voice and data transmission are of different characteristics than that of wireline. Problems encountered with radio systems involve geography, fading, rain attenuation, tower economics, etc., and were not emphasized in this study. Principal emphasis

"THE BALANCE OF THIS SHEET INTENTIONALLY LEFT BLANK" for this study was focused on transmission wireline facilities which are characterized as an unconditioned Type 3002 voice channel, Interstate Tariff FCC No. 260 circuits.

This type of voice channel is analog and exhibits limiting characteristics to transmission of relatively high data rates. Satisfactory performance of the voice/data transmission over this channel offers considerable economic benefits, since the costs associated with line conditioning (e.g., C1 or C2 conditioning) can be disregarded, but these costs are traded off in the form of frequency roll-off, noise, phase and amplitude equalization, etc. However, since Type 3002 unconditioned channels are currently used in abundance in present applications, successful application of voice/data equipment, in this environment, represents a viable near and intermediate term solution.

Longer range solutions require assessment of the technique for application in a digital transmission environment (such as that derived from T-1 type transmission facilities. Since this study scope precluded empirical measurements of a digital transmission configuration, assessment of applying the technique to a T-1 type transmission channel could only be performed in an analytical manner, and is covered in paragraph 2.8 of this report.

Various line characteristics associated with the unconditioned 3002 line, as well as any transmission line, are described and discussed in subsequent paragraphs. These characteristics have an effect on the voice and data signals in any transmission system. It was not possible, in the laboratory test, to cover all the effects that these line characteristics would have on the slot data signal. Consequently, these characteristics are defined in "text book" form as an aid in determining the effects that each would have on the data slot transmission system.

2.7.1 KEY CIRCUIT CHARACTERISTICS

Various circuit parameters that are of importance in evaluating channels for data transmission are identified below. Emphasis is primarily on voice-frequency circuits where most interest and activity is concentrated.

Some of the most important line characteristics which affect its suitability for data transmission are defined. The first four are important to voice communications as well, but become much more critical for data transmission and must be kept within tighter tolerances.

- LOSS - Loss of circuit is commonly measured at a frequency of 1000 Hz on an end-to-end basis with both ends of the circuit terminated in the proper impedance, usually 600 ohms.
- LOSS-FREQUENCY CHARACTERISTIC - This characteristic describes the variation of circuit loss as a function of frequency.
- RETURN LOSS - The return loss of a circuit is a measure of the energy reflection due to impedance mismatches. It is the ratio of reflected to incident voltage. Expressed in dB, return loss is equal to

$$20 \text{ Log } \frac{Z_0 + Z_R}{Z_0 - Z_R}$$

where Z_0 equals line characteristic impedance and Z_R equals terminating impedance.

For voice transmission the so-called "talker-echo," that is, the reflection back to the transmitting end of the circuit, is troublesome. For data transmission, however, the so-called "listener-echo" is of primary importance. The listener echo arrives at the receiver input, delayed in time, and interferes with the desired received data signal.

- STEADY NOISE - Steady noise consists of randomly phased components with energy uniformly distributed in frequency. A high level of steady noise, although undesired in voice transmission, reduces the reliability of data transmission in that it causes interference with the received level.

- IMPULSE NOISE - Impulse noise is probably best described in terms of sharp "clicks" or "bursts," and is commonly caused by switching transients of relays and the like. A single noise impulse can obliterate several bits of a relatively high-speed data signal.

- ENVELOPE DELAY - Envelope delay is defined rigorously as the rate of change of phase with respect to frequency. In the "real-world" this absolute quantity is of little importance; it is relative envelope delay - the difference between the envelope delay at a given frequency and the envelope delay at a reference frequency - which is the parameter of interest in data transmission. "Envelope delay" and "relative envelope delay" are the popular terms which have evolved from the more rigorously correct term "relative envelope delay." The effect of relative envelope delay on an information bit is to smear the bit in time, thus increasing the likelihood of interference between adjacent bits.

- FREQUENCY TRANSLATION - Frequency translation is the difference between the frequency of the transmitted and received signal. (It is commonly caused by a lack of synchronization between carrier reinsertion oscillators of single-sideband, frequency-division multiplex carrier systems.) It is detrimental to data transmission in that it changes the harmonic relationship of carrier and sidebands of the signal.

- PHASE JITTER - Incidental FM of the received signal, a rapid variation of the instantaneous phase of the received signal, is commonly called phase jitter. It is caused by noise or power supply ripple in carrier systems. PCM systems which quantize the analog signal prior to encoding and transmission can also generate phase jitter at certain input frequencies.

- PHASE HITS - Phase hits are relatively large, abrupt changes in phase of the received signal. They are caused most commonly by switching to alternate carrier supplies that are not in phase, or to alternate transmission facilities that exhibit different propogation delays. A single phase hit can at times obliterate a number of hits on the line.

- GAIN HITS - Gain hits are abrupt changes in circuit loss. Causes most effects in AM systems.

- DROPOUTS - Dropouts are short duration losses of signal. Probably causes are temporary open-circuit or short-circuit conditions on the line. Can cause loss of synchronization in a data circuit.

- NON-LINEAR DISTORTION - Non-linear distortion is caused by variations in circuit loss as a function of the instantaneous signal amplitude. Typical effects on voice and data signals are harmonic distortion, intermodulation distortion and, if particularly severe, clipping.

As previously stated, these line characteristics effect voice and/or data signals in some manner. It would be the objective of the system design to keep these effects to a minimum. In the case of the slot data transmission, small amounts of signal imperfections, caused by the line characteristics, can cause multiple error counts, primarily because the data transmitted is located within such a narrow bandwidth. To determine the absolute effects that these parameters would have on the data, would require extensive tests beyond the laboratory area; therefore, tests encompassing all line imperfections were not performed on this program.

2.7.2 OTHER TRANSMISSION SYSTEMS CONSIDERATIONS

In addition to the line characteristics listed in the above paragraphs that can cause imperfections in the transmission media, other interactions to the signal are encountered when interfaced with the transmission line equipment. These equipments are primarily "line repeaters" and "terminal" equipment. The terminal equipments are normally controlled by the system designer and are not discussed here; however, the repeaters are normally supplied by the transmission leasing company, i.e., "The Bell System," and are therefore not necessarily controlled by the leaser of the transmission facility. These repeaters are of two types, i.e., analog and digital.

A digital repeater is basically a regenerative repeater whose function is to reconstruct the digital signal. They are spaced at regular intervals along a digital transmission line and consist of three functional parts; namely, an amplifier equalizer, a regenerator and a timing circuit. The amplifier-equalizer shapes the incoming signal and increases its power level to aid the regenerator in making a pulse, no pulse decision, while the timing circuit provides timing information for the regenerator in order to minimize the chances of errors.

These devices, as the name implies, are of a digital nature, and would not be expected to be interfaced with slot study circuits. The digital signals in the slot study are converted by modulation techniques to an analog form before transmission, thus the digital repeater will not be discussed any further. Suffice to say they do exist and perform a useful function in the transmission of digital data in digital form.

In an analog system, the repeaters function is to match as closely as possible the nominal loss of the associated wireline. In order to perform this task, active devices are incorporated into the repeater design. As a result, repeater performance is characterized by parameters such as gain, noise figure, overload point and non-linear distortion coefficients. These characteristics which are inherent in the repeater design, effect the transmitted signals. The effects encountered have different interactions on the signal. For example, in an FSK system, high harmonic distortion would degrade a signal more so than in a PSK system; whereas phase distortion in the repeater would be more detrimental to a PSK system. Additionally, the load capacity can affect all types of modulation methods if the signal power affects the linear power properties of the repeater, thereby causing an "overload." This condition not only can cause damage to the repeater, but also change modulation indices by some prescribed amount. Similiarly, non-linear distortion, again primarily controlled by the active devices inherent in the repeater, can also effect the modulation properties of the transmitted signals.

As a result of these inherent properties found in the repeater equipment, the data as well as the speech signals transmitted cannot be recovered as efficiently as if this equipment did not exist. Conversely, without them, the transmission of signals would be limited to relatively short distances. Consequently, since these repeaters do make up part of the transmission facility, system designers must include them in their system performance specification.

In the case of the slot study, tests on repeater action could not be performed since only laboratory tests were included as part of the program. It is expected, however, that with the modulation technique (PSK) selected for the study, only a minimum effect would be encountered because of the repeater. Unfortunately, no data could be obtained during the course of the program, as indicated above, and therefore it is recommended that further tests involving repeaters be performed.

2.7.3 TRANSMISSION SYSTEMS COMPARISONS

One of the requirements of this study program was to transmit the slot data over an unconditioned type 3002 transmission line, Interstate Tariff No. 260 circuits. Additionally, as part of the study program, it was required that other lines, including conditioned lines, under Tariff No. 260,be investigated in order to determine if their characteristics would improve or hinder the transmission of the slot data.

As a result of this requirement, the subsequent information is provided covering the various transmission wireline facilities which can be used as an aid in formulating system design parameters.

2.7.3.1 INTERSTATE TARIFF FCC NO. 260 CIRCUITS

Under Tariff No. 260, a broad range of circuits are available for private line application. A summary of these offerings is given in Table 2.7-1. These circuits are derived from transmission facilities such as that described above, and in certain cases additional conditioning is required to provide the necessary channel characteristics for the intended application.

Of particular interest, consistent with the objectives of this study, are the Type series 2000 and 3000 channels which are intended for voice and alternate voice/data application, respectively. Table 2.7-2 identifies the circuit characteristics of the series 3002 channel with and without conditioning.

2.7.3.2 TYPICAL SUBSCRIBER SERVICE REQUIREMENTS

Subscriber requirements for communicating vary over a wide range of demand. It can be a common analog voice need, a sophisticated computer-to-computer connection, or a number of other conversational or message call arrangements. A summary table of typical service requirements is listed in Table 2.7-3.

In commercial practices, satisfying most of these services requires adaption of existing telecommunication networks to data transmission and also requires a specialized terminal arrangement. This arrangement in switched network application generally consists of a data modem and transfer controls, permitting the using station to transfer its operational mode from conversational to data. In general, such terminals are designed for up to relatively high speeds (5400 bps). However, extremely high speed data transfer rates cannot be served by this class of subset, principally because of the modem, and secondly because of the bandwidth limitation of existing switched network transmission facilities. For this reason, high speed transmission is normally provided over dedicated or leased facilities which do not enter the switched network but are point-to-point connections between end stations.

Of particular interest, relative to this study, is the error rate objectives indicated for these classes of service.

TABLE 2.7-1
Summary of Commercially Available Leased Facilities as Offered by Common Carriers Per Tariff FCC #260

TYPE SERIES 1000 CHANNELS

Unconditioned channels suitable for D-C mark space or binary transmissions at rates up to 150 baud. They may be physical circuits or derived from VFTG systems. Not intended to carry tone signals to any frequency.

Type	Application
1002	For transmission of 55 baud teletypewriter signals
1005	For transmission of 75 baud teletypewriter signals
1006	For transmission of 150 baud teletypewriter signals

TYPE SERIES 2000 CHANNELS

These channels intended for voice transmission only.

TYPE SERIES 3000 CHANNELS

Voice band channels to be used for transmission of data or alternate voice/data. Most commonly used leased data channels, serving as single channels from 150 to 9600 bps or subdividing into multiple low-speed telegraph channels.

Type	Application
3001	For remote metering, supervisory control and miscellaneous signalling.
3002	For data or alternate voice/data use with an approximate bandwidth from 300 - 3000 Hz.

TYPE SERIES 4000 CHANNELS

Voice band channels for transmission of telephotograph or facsimile signals. Special conditioning is applied.

TYPE SERIES 5000 CHANNELS

Telpack C & D Channels

Type	Bandwidth	Equiv. Voice Grade Channels	Application
5701	48 KHz	12	10 Hz to 20 KHz analog or 40.8/50 K Bps digital
5703	15 KHz	6	2 Level facsimile or 19.2 K Bps data
5751	240 KHz	60	200 Hz to 100 KHz analog or 230.4/250 K Bps data
5102		1/6	55 baud TTY
5105		1/6	75 baud TTY
5106		1/3	150 baud TTY
5302	3 KHz	1	All voice band data equivalent to 3002

TABLE 2.7-1 (Continued)

TYPE SERIES 6000 CHANNELS
For transmission of audio program material to AM, FM, or TV studios or stations, etc.

TYPE SERIES 7000 CHANNELS
For the transmission of video and ETV material.

TYPE SERIES 8000 CHANNELS
Channels for 50 K Bps data or a bundle of 12 channels of voice circuits.

Type	Application
8801	A bandwidth of approximately 10 Hz to 20 KHz or 40.8 K Bps or 50 K Bps
8802	Same as 5000 series. This 12 channel "bundle" can be extended beyond terminal point with restriction of only one terminal at this point.
8803	For facsimile in the range of 29 - 44 KHz or 19.2 K Bps data

TYPE SERIES 10000 CHANNELS

For interconnection to private microwave systems. Common carrier will meet a private microwave system at distances of up to 25 miles from a privately-owned branch exchange and interconnect to a switchboard.

TYPE SERIES 11000 CHANNELS
Offering of 12 or 60 voice grade channels or a facility for 50 K Bps or 240 K Bps data service. Available for user sharing of service/bandwidth.

CONDITIONING
For Type 3002, 5302 or 8302 channel phase and amplitude equalization, plus impulse noise control for making channels suitable for moderate to high data speeds.

C1 - Two point or multipoint
C2 - Two point or multipoint
C4 - Two point or three point

TABLE 2.7-2 (27)

	CIRCUIT TYPE							
Circuit Designation	3002		C1		C2		C4	
(Noted D & G)	← Alternate Voice/Data or Data Only →							
General Characteristics								
1) Type of Service	← Point-to-Point or Multipoint →							
2) Mode of Operation	← Simplex, Half or Full Duplex →							
3) Method of Termination	← 2W or 4W →							
4) Impedance (Source & Load)	← 600 ohms Resistive Balance →							
5) Max. Sig. Pwr. (Note H)	← 0 dBm for Composite Data Signal, 0 VU for Voice. →							
Attenuation Characteristics								
1) Meas. between 600 ohm impedances at lineup (Recommended)	← 16 dB ± 1 at 1000 Hz →							
2) Expected max. variance of loss (Note A)	← Short Term ± 3 dB; Long Term ± 4 dB →							
3) Frequency Response (Ref. 1000 Hz) (Note B)	Freq. Range (Hz)	Variation (dB)	Freq. Range (Hz)	Variation (dB)	Freq. Range (Hz)	Variation (dB)	Freq. Range (Hz)	Variation (dB)
	300-3000	-3to+12	300-2700	-2to+6	300-3000	-2to+6	300-3200	-2to+6
	500-2500	-2to+8	1000-2400	-1to+3	500-2800	-1to+3	500-3000	-2to+3
			2700-3000	-3to+12				
4) Frequency Error	← ± 5 Hz →							
Delay Characteristics								
1) Absolute Delay (Note C)	← NOT SPECIFIED →							
2) Envelope Delay Distortion	Freq. Range (Hz)	Delay (μsec)	Freq. Range (Hz)	Delay (μsec)	Freq. Range (Hz)	Delay (μsec)	Freq. Range (Hz)	Delay (μsec)
	800-2600	<1750	1000-2400	< 1000	1000-2600	< 500	1000-2600	< 300
			800-2600	< 1750	600-2600	< 1500	800-2800	< 500
					500-2800	< 3000	600-3000	< 1500
							500-3000	< 3000
Noise Characteristics								
1) Message Circuit Noise	← SEE TABLE (Page 111) →							
2) Impulse Noise (Notes E & H)	← 15 counts in 15 minutes at 69 dBm VB (69 dBm C) →							

TABLE 2.7-2 (Continued)

NOTES

A - (L) is the net loss as measured at 1000 Hz. Short term variations are those likely to be observed during a measurement interval. They are caused by amplitude and phase hits, dropouts and maintenance activities. Long-term variations include seasonal changes, component aging, etc.

B - DC continuity is not provided on any of these offerings.

C - Absolute delay and propogation times are not specified. Where satellite channels are employed, the delay may be several tenths of a second and telemetry and retransmission schemes may be either unusable or limited.

D - If alternate voice data operation is desired and the data modulation does not allow the use of companders (such as many AM systems where instantaneous power varies rapidly), the voice mode may be degraded by excessive noise. If signalling is required, the data modulation must not interfere with 2600 Hz S.F. signalling units and response is not specified between 2450 Hz and 2750 Hz.

E - These impulse noise limits are primarily Plant Maintenance Limits. In cases where they are exceeded, engineering will evaluate the performance on impulse noise distribution, i.e., how rapidly the counts (impulses) fall off as counting level (impulse noise peak voltage) is raised, and the effect on the data system performance.

F - Third-point operation describes the conditioning where point A (master) can transmit to B and C (slaves) simultaneously, and both B and C can respond to A. Transmissions between B and C are possible, but the characteristics are not specified.

G - C3 conditioning not included in this table, describes conditioning of access lines and trunks in central office switching applications. An end-to-end connection, consisting of four trunks and two access lines with C3 will approximate C2 conditioning overall.

H - The "VB" in the objectives refers to the voiceband filter in the measuring set. This approximates the "C" message filter and the typical response of the voice grade channel.

MESSAGE CIRCUIT NOISE CHARACTERISTICS PRIVATE LINE OPERATION TABLE

The basic objectives for data operation is that the rms data level should be 24 dB above the message circuit noise reading with a "C MESSAGE WEIGHTED FILTER" during the data "signal-on" condition. Since the data level can be -16 dBm at the terminal, the following objective is given:

*rms data signal	-16 dBm	74 dBrnC
Signal-to-noise requirement	24	24
Allowable noise	-40 dBm	50 dBrnC

If readings are made in idle or "no signal" condition, as most common, the following objectives are typically used. These readings are caused by a masking of noise, due to idle circuit loss in the expandor in companded carrier systems, and other effects.

Circuit Length (Miles)	Expected Noise Reading C Message Weighting Not Exceeded (dBrnC)
0 - 50	28
50 - 100	31
100 - 400	34
400 - 1000	38
1000 - 1500	40
1500 - 2500	42
2500 - 4000	44
4000 - 8000	**47
8000 - 16000	**50

* Assumes a random spectrum of the data signal
** Voice operation may be degraded

Note: All readings are expected values. While the noise characteristics are fairly staple, variations due to facility activity or troubles will be experienced.

TABLE 2.7-3 [28]

SUMMARY OF TYPICAL SUBSCRIBER SERVICE
REQUIREMENTS AND ERROR RATE OBJECTIVES

Subscriber Service	Nominal Frequency Range/Data Rate	Objectives for Minimum Permissible Error Rate
Analog Voice	300 - 3000 Hz	
Digital Voice	2400 bps	10^{-4}
Digital Voice	9600 bps	10^{-4}
Digital Voice	50 KBps	10^{-3}
TTY Terminals	Up to 150 baud	10^{-5}
Data Terminals	150, 300, 600, 1200, 2400, 4800 bps	10^{-5}
Facsimile	2400 bps	10^{-5}
Facsimile	50 KBps	10^{-4}
Computer	(Note 1)	(Note 1)
Video Phone	4.9 MHz	10^{-5}
Digital TV	39.3 MHz	10^{-5}

Note 1 - Transmission transfer rates vary from low-speed to very high speed (megabit). For very high rates either parallel or slow-down transfer techniques should be considered, consistent with the accessed network's capability and capability to handle the traffic.

2.7.4 SECTION 2.7 - SUMMARY & CONCLUSIONS

Various transmission systems have been examined and key circuit characteristics identified.

Commercially available private line leased services have also been examined and tabulated.

Additionally, private line characteristics of unconditioned and conditioned lines have been compared, indicating the variation in characteristics principally associated with the envelope delay distortion and frequency response parameters.

Typical subscriber service requirements have been identified with corresponding error rate objectives. It should be recognized that these are user objectives for desired service, and not anticipated service quality of transmission facilities. These objectives are generally attained through the use of conditioned lines coupled with some method of error detection and correction. Pertinent to this study is the observation that these error rates, being state-of-the-art objectives, are comparable to that anticipated for application to the Air Traffic Control requirements.

However, it should be stated that this section (2.7), in its entirety, indicates that the various transmission systems, including their respective equipments, e.g., repeaters, will effect the slot data transmission in one form or another.

Therefore, in order to ascertain system efficiency, it is recommended that tests be performed over actual line conditions, the resulting data analyzed with respect to the data slot transmission technique, and parameters be defined prior to any system implementation.

2.8 INTERFACE WITH T1

The term T1 line refers to a digital transmission line operating at 1.544 megabits per second. It is one member of a hierarchy of digital transmission lines which carry digital pulse information. Digital transmission offers advantages over analog transmission. In analog transmission disturbances, once introduced, cannot be eliminated and information content is degraded. Degradation accumulates and ultimately sets a limit to system performance. In digital transmission, the transmission system does not alter the information content until the degradation becomes so severe that the receiving equipment misreads pulse and no pulse conditions. Below this threshold level, the information transfer is essentially perfect. Also, digital transmission is not limited by accumulated degradation since the signal can be regenerated (read and retransmitted) before degradation becomes severe.

To exploit the advantages of digital transmission, it is necessary to convert the analog information into digital form for transmission and to reconvert the digital information back to its primary form at the receiver. The required conversion is performed at the terminal equipment. At the terminal equipment, the analog signal is sampled, quantized, and pulse code modulated to produce a digital code representation of the analog signal for transmission. At the receiving end, the code pattern is decoded to produce a voltage proportional to the original analog sample. Two encoding schemes are in use in T1 channel bank (terminal) equipment. The older D1 channel banks intended for subscriber and direct trunk service utilize a 7 bit code of 128 steps, while the newer D2 and D3 equipment utilize an 8 bit code of 256 steps.

The encoding scheme used in both types of channel bank equipment is designed to encode voice signals. Studies of speech signals has shown that the distribution of amplitudes is not uniform, and that for a given talker volume smaller amplitudes are more probable than larger amplitudes. A better signal to distortion ratio can be expected if more attention is paid to low level signals,and the error characteristic is made smaller for the more probable amplitudes at the expense of larger errors for the less probable amplitudes. Secondly, speech signals have a dynamic range of up to 40 dB. With uniform encoding, weak signals will experience 40 dB poorer signal to distortion ratio than strong signals. The signal to quantizing distortion ratio (S/D) is generally the chief noise contributor in a properly operating PCM system, and therefore great improvements in signal to noise performance are obtained by properly optimizing the quantizing steps.

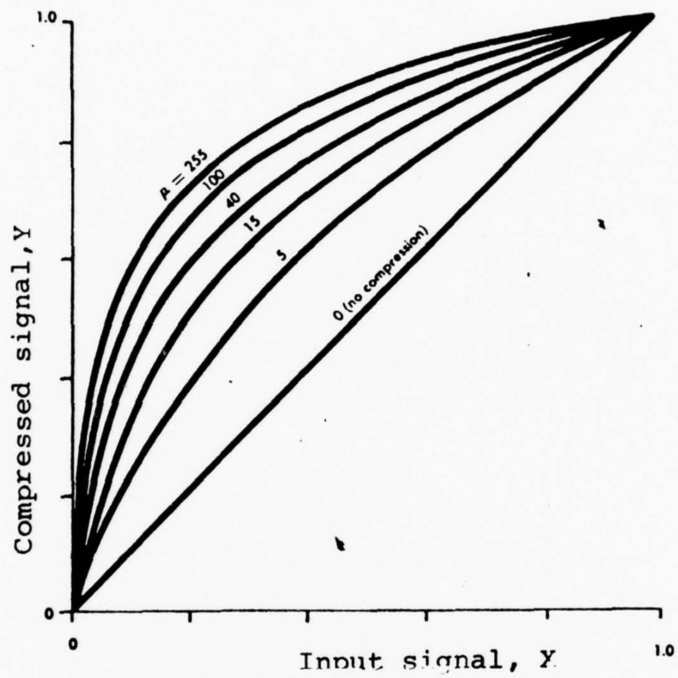
In the D2 channel bank a non-linear coder is employed which follows a mathematical law selected to approximate a constant signal to noise ratio. Compression and encoding functions take place simultaneously in the coder which follows the equation:

$$Y = \operatorname{sgn}(x) \frac{\ln(1+\mu|x|)}{\ln(1+\mu)} \qquad -1 \leq x \leq 1$$

where X and Y are the input and output voltages , $\ln$ is the natural logarithm, and μ is a constant.

The quantizing characteristic for various values of μ is illustrated in Figure 2.8-1.

The value of μ is 255 for the D2 channel bank. In contrast, the D1 terminal employs a μ value of 100. Also, the quantizing characteristic is often called the companding characteristic since it is effected through the combination of an instantaneous compressor-expander (compander) and linear coder-decoder rather



Logarithmic compression characteristics.

Figure 2.8-1

than the non-linear coder-decoder employed in the D2 system.

In effect, the non-linear encoder measures the voice signals using a variable increment. Low levels are measured with a finer resolution than higher level signals. For the D2 channel bank equipment, the measurement increment varies from about a quarter milli-volt for signals below -45 dBm0 to thirty-five milli-volts for signals just below the 3 dBm0 overload point. In actual practice, the μ law requirement is met by a piecewise linear approximation. In the D2 channel bank approximation, each half of the curve is divided into 8 segments or chord of 16 steps each. The step size within each chord is constant. The basic characteristics of the D2 channel bank encoder-decoder are listed in Table 2.8-1.

Since the non-linear encoder-decoder is designed to process voice information, it is important to analyze the effect of the encoder-decoder processing a signal consisting of voice mixed with a data tone.

As shown in Figure 2.8-2, the complex voice signal and data tone are added to produce the combined voice with data signal. As can be seen in the figure, the combined voice-data signal spans a wide dynamic range. The voice and data will interact with each other to degrade the transmission of both voice and data. Detail Figure 2.8-2 (Sheet 2) shows the effect of data on the voice signal. The low level voice signals (which are the most important for voice quality) ride on the data tone.

CHORD	STEP SIZE AS A % OF FULL SCALE	CHORD END-POINTS AS A % OF FULL SCALE	CHORD ENDPOINT IN DB DOWN FROM FULL SCALE
0	0.025%	0.37%	-48.55
1	0.050%	1.16%	-38.73
2	0.10%	2.73%	-31.29
3	0.20%	5.86%	-24.63
4	0.40%	12.1%	-18.32
5	0.80%	24.7%	-12.15
6	1.60%	49.8%	-6.06
7	3.20%	100%	0

μ LAW (255) ENCODE-DECODE CHARACTERISTICS

Table 2.8-1

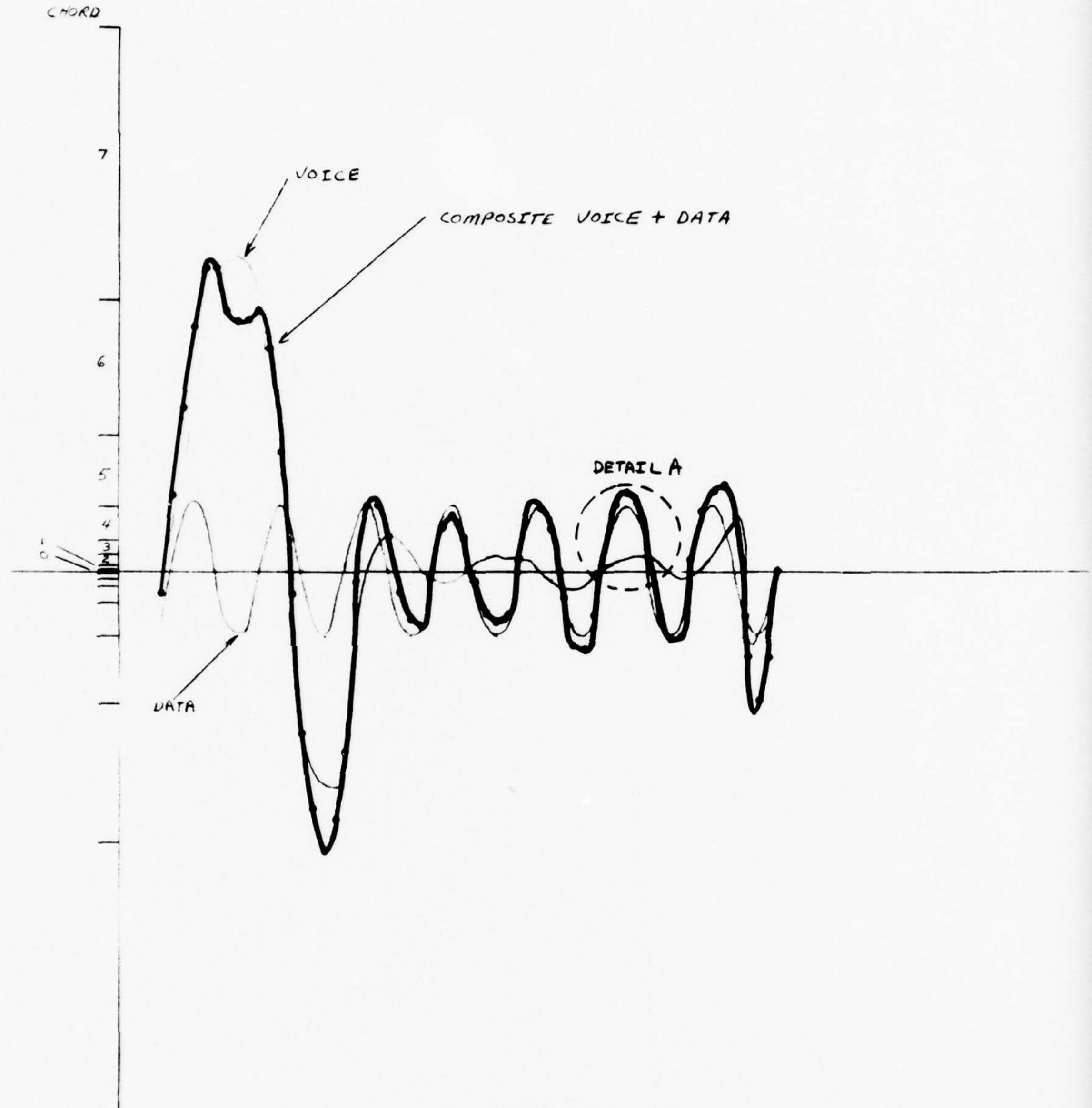


Figure 2.8-2

Sheet 1 of 2

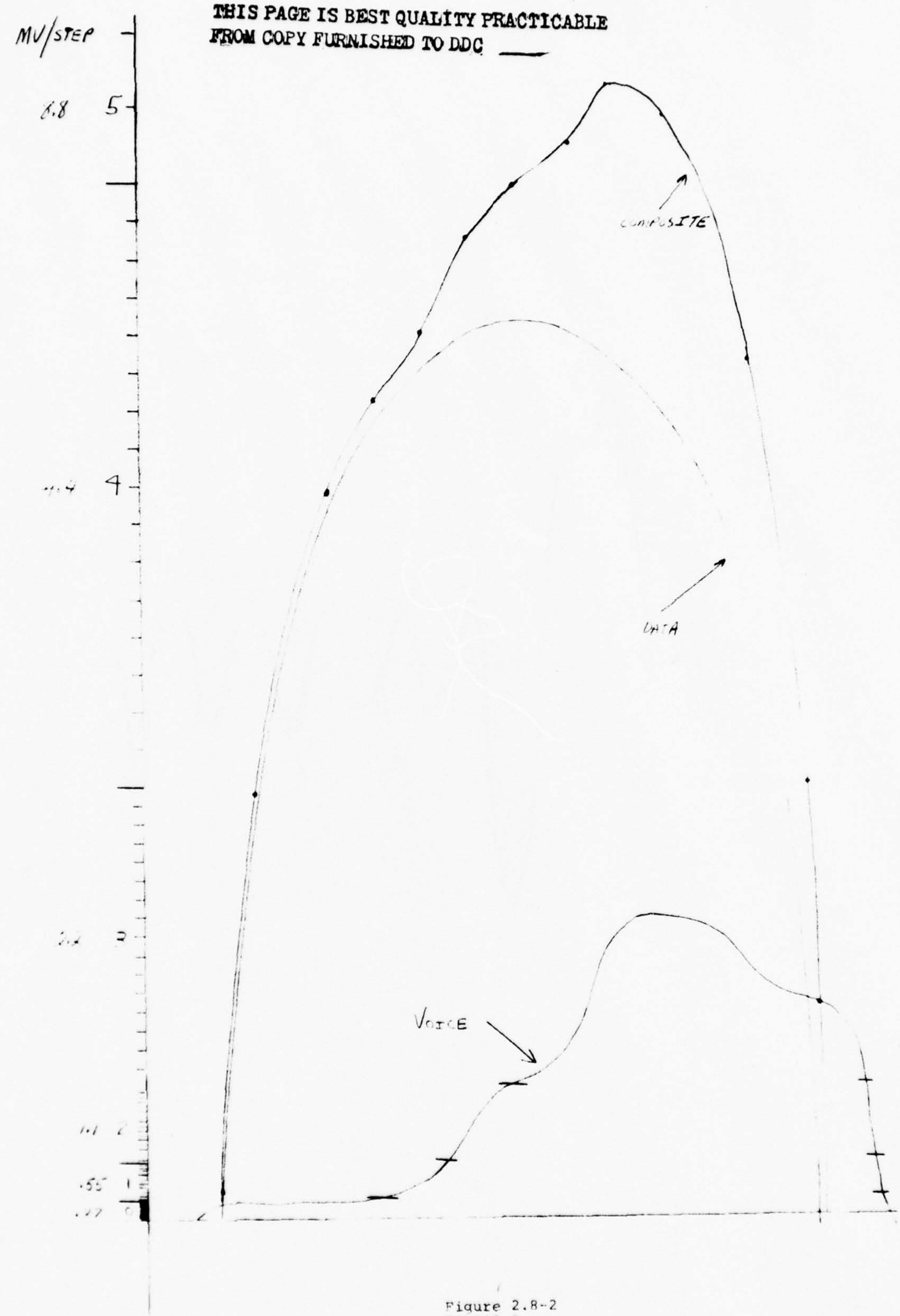


Figure 2.8-2
Detail A

Sheet 2 of 2

This causes the low level signals to be quantized at a higher level than they would if there was no data tone present.

With no data present, voice would be quantized within chords 0 thru 3. In chord 0, the step size is .27 milli-volts and it doubles in each chord up to a value of 2.2 milli-volts in chord 3. When this same voice energy is added to the data tone, quantizing takes place at higher levels. In this example largely in chords 4 and 5.

The effect of this higher level quantization is to reduce the signal to distortion ratio and increase channel noise. The amount of noise increase is difficult to predict. It depends on the relative levels of voice and data. As a first order approximation, we can consider the voice as being elevated by the RMS value of the data tone. The companding characteristic is such as to maintain signal to distortion constant. Thus considering each component transmitted separately (voice and data tone)

$$\frac{S_V}{D_V} = \frac{S_T}{D_T} = C$$

where C is approximately 35 dB for the D2 channel bank. When the voice signal is elevated, the signal is quantized at the tone distortion.

$$\frac{S_V}{D_T} = \frac{S_V}{S_T} = C$$

Since we are considering low levels of voice $S_t > S_v$ so that in dB:

$$\frac{S_V}{D_T} = C_{DB} - (S_{T_{DB}} - S_{V_{DB}})$$

Thus the voice signal to distortion ratio is reduced by level difference between the data tone and voice. With a -20 dBmo date tone, voice signals of -40 dBmo will have the signal to distortion ratio degraded by 20 dB. The analysis is a gross approximation, however, it indicates that significant voice distortion will be encountered when the composite voice and tone is passed through a voice compander.

Considering only this one aspect of voice-data transmission, the data level should be kept much lower than the voice level. Figure 2.8-2 also shows the effect of voice on the data signal. At times the low level data signals ride on a voice peak. This causes the low level data signal to be quantized at a higher level than if there were no voice present. If the data signal is kept at a low level, as required to minimize voice distortion, data signal distortions, with possible loss of information, can be expected on the high voice peaks.

It is felt that the encoding-decoding operation will present the greatest problem when operating with an inband signalling tone on PCM carrier equipment. Distortion and noise which result because of errors in the digital signal stream will be minor when compared to the signal perturbations which are injected by the encoding-decoding with companding function.

Prior to any future system design, actual measurements should be made by subjecting the composite voice-data tone to the PCM encoding-decoding function. Since D1 type equipment is of an older design and is not being used on new installations, the measurement need only be made on D2 type equipment.

One possibility for transmitting digital information with voice over a T1 link is to use the normal carrier signalling path. In the D2 channel bank equipment every sixth frame carries signalling information. With a frame rate of 8 kHz, there results a signalling data rate of 1333 Hz. Channel bank equipment would have to be modified to accept asynchronous digital data on the signalling path input and synchronize the data to the T1 link clock. However, it appears possible to make a modification and utilize the T1 equipment.

2.9 INTERMODULATION RESTRAINTS

Any electrical network, to some degree, is subject to non-linearities. While good design practices tend to reduce the undesirable effects caused by these non-linearities, they are from a practical standpoint impossible to eliminate completely. This results primarily because non-linear elements are incorporated in the network, both in the transmission media, e.g., repeaters, and in the terminal equipment. These elements include diodes, transistors, other active devices, as well as some passive devices such as transformers which utilize ferrous materials. Of the various types of non-linearities introduced by these devices, one of concern is that of intermodulation distortion or noise. In the case of the data slot study, care was exercised in keeping these intermodulation signals to a minimum by ensuring that amplifiers, buffers, line drivers, etc., were provided with signals that would not saturate the active device, and that the transformer would not be provided with signals which would cause core saturation. Additionally, the signals connected to the artificial line, during the laboratory tests, were within the range of the equipment to lessen the probability of distortion in this device. However, because field tests were not conducted on the circuits over an actual line, it is not possible to determine the effect of the transmission media on the circuits in the "real world," but recommendations can be made. Since most of the non-linearities are caused by overloading, it is recommended that the transmitted signals be kept within the bounds of the transmission system specified by the "common carrier" company. Additionally, it is recommended that the circuit active elements also be kept within the manufacturers tolerances. In this manner, the power of these non-linear signals are kept to a minimum, which in turn reduce the probability of errors in the data signal and increase the quality of the voice signal.

2.10 CODED AND UNCODED DATA

Over the past twenty-five years there has been a tremendous expansion of data communications. A great deal of effort has been expended on the development of techniques to assure accurate data transfer. The various techniques used to reduce errors all have at least one thing in common, they all attempt to minimize the effect of noise in one way or another. Filters and modulation do this by selectively removing the noise. Error control does it by attempting to average the large number of good bits, resulting from low-noise transmission against the relatively few expected bad bits.

Tests carried out by others using a frequency modulation system, indicated that successful transmission in the tested mode could be achieved on a large majority of communication lines at up to 1200 bits per second using only a compromise equalizer to correct the more usual forms of line distortion.

These test results, as reported by Alexander, Gryb and Nast, can be summarized as follows:

- Error rates are better on low loss circuits than on high loss ones, and therefore generally better on shorter circuits.
- With non-zero probability, a circuit with a nearly arbitrarily high error rate can be found, but circuits with fewer than one (1) error in 10,000 bits are far more common. Over half of the circuits yielded fewer than one error in 100,000 bits.
- Somewhat poorer error rates are achieved at higher transmission rates, but errors are not markedly worse until line distortions nearly destroy transmission by means other than noise.

- With reasonable error control schemes the undetected error performance can be improved to something on the order of "1" error in 10^9 bits transmitted.
- Most errors are single isolated errors.
- A sufficient number of multiple errors occur so that single-error control schemes do not improve the error rate by even as much as a factor of ten (10).
- Error detection with retransmission is probably more efficient in most cases and is almost certainly less expensive.

Later tests conducted by Townsend-Watts and Balkovic essentially confirmed these results. They also showed an improvement in telephone plant in the 1960 to 1970 time frame between tests. However, even with improved plant, it may be inferred from the above test results that some form of error control is necessary where accuracy of data transmission is an important requirement. Therefore, an analysis of popular existing error control techniques is worthwhile in order to determine applicability to the slot signalling problem.

2.10.1 ERROR CONTROL TECHNIQUES

Two well-known techniques are used to provide an error control capability. These are Forward Error Correction (FEC), and Error Detection with Automatic Retransmission (ARQ) upon receipt of a retransmission request from the receiving end of the circuit.

Both of these error control techniques require the generation of redundant error check bits at the transmitting terminal. The redundant bits are combined during transmission with the original data.

With FEC, the redundant bits are used at the receiving terminals to detect, locate, and correct any transmission errors before delivery to the data sink. With ARQ, the redundant bits are used only for the purpose of detecting errors in the received data.

A comparison of FEC with ARQ techniques leads to the following general conclusions:

. <u>FEC KEY ADVANTAGES</u>

  . Does not require a feedback channel and can therefore be used in oneway transmission systems.

  . Can be used efficiently in broadcast applications.

  . Does not require temporary storage of data for retransmission purposes.

. <u>FEC KEY DISADVANTAGES</u>

  . Error control codes are highly dependent upon the statistical error characteristics of the channel. If errors occurring in the channel exceed the error correcting capability of the code, erroneous data will be delivered to the data sink.

  . Codes require a large number of redundant bits.

  . In general, cost and complexity requirements exceed that for ARQ systems.

. <u>ARQ KEY ADVANTAGES</u>

  . Certain ARQ systems can provide a high degree of accuracy for transmitted information regardless of actual channel error statistics. Error detecting codes exist that can provide extremely low undetected message error rates even when bit error probabilities approaching 1/2 exist.

  . Error detecting codes do not require a large number of redundant bits, therefore, ARQ systems use the data channel more efficiently than do FEC systems.

  . In general, cost and complexity requirements are less than that for FEC systems.

- <u>ARQ KEY DISADVANTAGES</u>

  - Requires temporary storage of data at both terminals for potential retransmission requests at the transmission end and total detection process at the receiving end.

  - Requires a feedback channel and, therefore, cannot be used for one-way transmission systems.

  - Cannot be used efficiently for broadcast networks.

The error control techniques described above require the use of error control codes. However, there is one other technique available for improving error rate performance of a channel that does not require the coding of information, and that is repetition.

A simple error control system can be obtained by transmitting the same message a number of times over one channel or simultaneously over a number of different channels. Errors can be detected by disagreements among the received messages. Also, errors may be corrected by a majority decision if more than two messages or channels are used. This technique is generally inefficient in its use of channel capacity. However, it is useful in systems for which simplicity of terminal logic circuitry is a priority criterion, and/or where efficiency of channel capacity is not the limiting factor.

The problem of selecting an effective error control system is not a simple one. It requires the consideration of many aspects of the overall system objectives. In particular, consideration must be given to system requirements such as: Message accuracy, tolerable delays, message format, and cost of implementation.

- MESSAGE ACCURACY - System requirements for message accuracy are generally stated in terms of an output error rate. This error rate is dependent upon the characteristics of the transmission channel and the particular error control equipment employed.

  Message accuracy is an extremely important consideration in potential applications of the slot signalling technique. Several types of data are likely to be transmitted. Data types include: radio keying signals, status change information such as main/standby switching or frequency selection, and alarm and status reporting.

  The above analysis of presently employed techniques indicates that extremely low error rates are possible. However, low error rates are acquired at the expense of hardware cost or channel capacity and message reception delays.

- TOLERABLE DELAYS - The process of error control introduces a delay between transmission and final reception of information that is exclusive of any propogation delay. In FEC systems the delay is fixed and the information is delivered to the data sink in a continuous stream of bits or characters. With ARQ, the delay is fixed during periods of good transmission and is variable during the periods of poor transmission when retransmissions are required. In ARQ systems, information is delivered to the data sink in bursts of bits or characters.

  Message transmission delay is also an extremely important consideration. It has been determined that a push-to-talk (PTT) command, and its associated confirmation response, should be transmitted with a round trip delay, due to message formatting encoding and modulation/demodulation of less than 80 ms. If we assume an equal allocation of delay between transmission of command and return of confirmation, the command must be transmitted within 40 ms. If the data rate is 200 bits per second, the push-to-talk command must be transmitted within 8 data bit times. This is a very small amount of time. Even if the delay specification is relaxed to 80 ms one way, only 16 data bit times are available for transmission.

  The delay requirement will most certainly rule out an ARQ system due to its increased potential for added message transmission delay, and it will be a very important consideration in the development of a potential technique in the following paragraphs.

- MESSAGE FORMAT - The grouping of characters into blocks can increase the probability of detecting transmission errors and can improve the efficiency of information transfer.

  Data format must be carefully organized to assure that accuracy and delay requirements are met. Format must assure accurate bit and character synchronization.

- COST has been and probably always will be an important consideration. However, the current availability of MSI and LSI circuit technology including RAMS, ROMS, parity trees, majority voters, and CRC polynomial generators has made many heretofore impractical error control techniques practical. Cost is not expected to be a problem if accuracy and tolerable delay problems can be solved.

2.10.2 DATA WORD FORMAT

The above discussion indicates that message delay will be a significant problem in a practical slot signalling approach. If we assume that push-to-talk information is transmitted in defined bit locations, the delay requirement and 200 bit per second data rate will limit data "block" or character size to a range between 8 and 16 bits.

If the system application requires independent selective keying on a specific channel, it will be necessary to provide a specific data bit for each key signal required. Bits will also be required to transmit status and alarm information.

Because of the relatively low data rate it will be assumed that data will be transmitted in an asynchronous format. With asynchronous transmission, a clock signal is not transmitted with the data bits and the data bits need not be contiguous. In order for the receiver to properly recover the message, bits are grouped into data characters and synchronizing start and stop elements are added to each character.

Figure 2.10-1 illustrates a potential data character format.

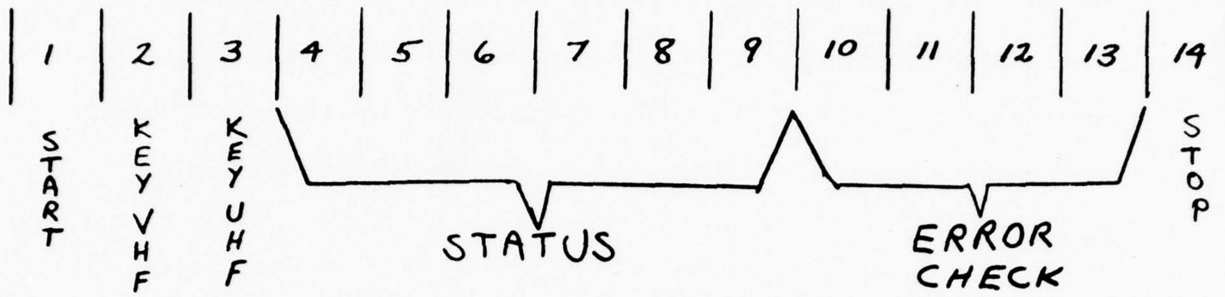


Figure 2.10-1

As can be seen in the figure, the basic character length is 14 bits. Two bits are allocated to key separate radio frequency bands. Six bits are allocated for the transmission of status information. Four error check/correction bits are provided. At this point an error correction code has not been specified, but for the purposes of this analysis, the four error control bits appear to be a practical choice.

If we assume that the 14 bit character is transmitted repeatedly, a worst case push-to-talk delay occurs under the following conditions:

a. A push-to-talk is activated immediately after bits 2 and 3 are transmitted, causing transmission of the push-to-talk to be delayed until the next character is transmitted.

b. Bits are not utilized by the receiver until a whole character has been inputted.

Under these conditions, push-to-talk transmission could be delayed 1 character plus 11 bit times, for a total of 25 bit times. This equates to a delay of 125 ms at a data rate of 200 bits per second. Even if we assume that the key bits are unprotected by the error check code, such that it is not necessary to wait until a whole character is received before using the key data, a delay of up to 14 bits or 70 ms is possible.

It is possible that the character length can be reduced. However, it is unlikely that reductions will be such that a 40 ms target for 1 way transmission will be reached with a 200 bit per second data rate. Also, in the above discussion, we have assumed no error in transmission or loss of receiver word synchronization. Additional time will be lost if the receiver must wait "until the next character" to detect a push-to-talk due to a data error or worse, although not as common, must wait until word synchronization is reestablished.

Knowing the approximate character size and format, it is possible to review various error correcting code schemes to determine their potential applicability to the slot signalling problem.

2.10.3 ERROR CONTROL CODES

The paragraphs below briefly describe several error control codes that can be used to implement a particular error control technique.

- ERROR CONTROL CODES FOR ERROR DETECTION

  - CONSTANT RATIO CODES - A constant ratio code is one for which the number of zeros (spaces) and ones (marks) in a binary code word is fixed. A code word, as used here, is a group of bits that represents a single alphanumeric character. Constant ratio codes can be used only for error detection of character oriented messages.

  - SIMPLE PARITY CHECK CODES - These codes are constructed of n information bits plus one parity (check) bit. The parity bit is selected to make the total number of ones in the code word odd or even.

  - GEOMETRIC CODES - These codes include horizontal and vertical parity checks, commonly called block parity, as well as more complex code structures.

  - BOSE-CHAUDHURI-HOCQUENGHEM CODES - BCH codes are cyclic error detecting and correcting codes. A BCH code length is generally equal to one less than two raised to any power ($2^n -1$). The number of information bits and the number of redundancy bits are related to this power. (Other BCH code lengths, not related to the power "n," do exist but are not known to be used in currently available equipment.

    For a given pattern of information bits in a BCH code, one and only one pattern of redundancy bits will satisfy the code. Since the bits are binary, there is a probability equal to the error rate that a redundancy bit is in an erroneous state ("one" or "zero"). Therefore, the probability of all redundancy bits occurring in an erroneous state by chance is the error rate raised to the number of redundancy bits power. For example, with the BCH (15,11) code, the probability of a random set of error bits (generated by noise) indicating a valid erroneous word is the error rate raised to the power of four (4). At an error rate of 1×10^{-2}, the probability of undetected error is (1×10^{-2}) or 10^{-8}. At 200 baud, an error would go undetected once every 86 days.

. <u>ERROR CONTROL CODES FOR FEC</u>

- <u>CONVOLUTIONAL CODING</u> - Convolutional codes, also called recurrent codes, are distinguished by the absence of block structure. Information bits and parity check bits are interleaved on the channel, each parity bit being derived from a fixed number of preceding bits.

- <u>DIFFUSE CONVOLUTIONAL CODING AND THRESHOLD DECODING</u> - The diffuse convolutional coding and threshold decoding technique is a modification of the convolutional coding technique. Basically, the diffuse coding technique consists of the diffusion, or spreading, of the information bits used in the generation of parity check symbols. The amount of diffusion is usually based on the longest fade or interference to be expected on the channel.

- <u>BOSE-CHAUDHURI-HOCQUENGHEM CODES</u> - The description and capabilities of the BCH codes for error detection has been explained in the ERROR DETECTION portion of this section. This class of code can also be applied to error correction, but its implementation is complex.

- <u>GOLAY CODE</u> - The Golay code is a cyclic block-code consisting of 12 information bits plus 11 redundant bits. This code is generally called the Golay (23,12) code, and is capable of correcting all patterns of three or fewer errors. Although other codes exist that can correct three errors or less, the Golay code requires the least number of redundant bits per information bits.

- <u>TIME SPREAD CODING</u> - Time spread coding is based on the simultaneous use of time diversity and error control block coding. The code design is based on the assumption that errors occur in bursts on real channels. In general, to correct error bursts using block codes alone requires the use of long codes. The objective of time spread coding is to make the error bursts look like random errors so that short block codes can be used to correct the errors.

The effectiveness of an error control system is usually measured in terms of how much the system improves the error rate and reduces the efficiency. Error detection systems capable of dealing with large numbers of consecutive errors may be obtained for an inexorbitant investment in circuit complexity. Without this investment, error correction systems are often out of the question because of the presence of long bursts of errors. Hence, there is a tendency in the digital communication field to provide error detection of a very high order and to correct the erroneous material by retransmission. This is most frequently done by making use of collections of interlaced parity groups. Such interlaced groups are separated by several bits in the message sequence. Providing this interlacing process is carried far enough, even completely catastrophic failures will still be detected with a very high probability, for the resulting random sequence of binary digits will be very unlikely to satisfy all of the various parity checks simultaneously.

Alexander, Gryb and Nast present a good deal of information relative to results of tests of transmission over telephone circuits, and the effectiveness of some of the more common codes when used in telephone circuit application.

Some of the facts which have been reported are the following:

- Simple parity checks in which the members of the group being checked are all in sequence in the message usually result in an improvement by a factor of no more than two to four, because error bursts cause a high probability of local error correlation.

- Systems of fixed count coding, in which the number of "ones" in a group is always constrained to be the same, are only marginally better than simple parity check systems.

- Recurrent burst-correcting codes (such as those originated by Hagelbarger) only improve the error rate by a little more.

. Suitably interlaced parity checking can be made as effective as desired, but the complexity of the equipment to instrument the process may rise to relatively high levels.

. A number of forms of interlaced code can be made similarly effective. Again, with a high cost level.

Thus, it appears that due to the stringent transmission delay requirements, ineffectiveness of error correction coding in a burst error environment, and small amount of data that need be transferred, error detection with repetitive transmission appears to be the most effective data protection approach. A Bose-Chaudhuri-Hocquenghem code which can be easily implemented with available monolithic CRC polynomial generators appears to be a good choice.

2.11 RELIABILITY

This study has demonstrated the feasibility of transmitting and receiving co-existant signals (voice and data) without degradation to either over the same transmission path. Speech and data signals can be inserted into an unconditioned voice grade line between the control centers and the remote air-to-ground radio sites, with the data using only a frequency slot in the voice bandwidth.

By utilizing error detection coding techniques, the probability of an undetected error can be made very low. The system can, therefore, be reliably used to transmit status type information. Additional tests will have to be made in the field to verify the characteristics of transmission facilities and to determine actual practical error rates including burst and impulse noise effects. From these measurements the amount of error protection required can be determined.

However, the process of error detection increases message transmission delay. A percentage of data bits are used in the data encoding process. Also, one must wait until a specified portion of the message is received prior to making a determination of accuracy. This increases transmission delay. Thus, it is felt that the use of the slot for transmission of push-to-talk information is questionable due to stringent delay requirements. Careful investigation should be made of any proposed system implementation with respect to the delay requirement.

2.12 OTHER SYSTEM CONSIDERATIONS

In configuring a system such as the one under investigation in this study, aside from the transmission facility's characteristics, there is an additional concern regarding the use of AGC amplifiers, and their effect on the filtered notch within the nominal voice band. Figure 2.12-1 illustrates this concern. In the figure a typical arrangement of the voice plus in-band data circuit shows the one way path (transmitting from a remote radio site to a ATC position, as an example). Voice reception by the radio receiver is transmitted to the ATC position via the MOD/DEMOD portion of the voice plus in-band circuit.

The data signal contribution to the speech plus data combined signal is injected into the circuit at point A.

The combined signal (voice plus data) is then transmitted toward the ATC center via the transmission circuit. In this transmission path both voice and data signals will experience normal propogation attenuation but will be restored to the desired level(s) by the RCV Line Amplifier (Point B). The output of the RCV Line Amplifier is injected into the ATC center voice plus in-band data channel terminal where data and voice signals are separated by appropriate filter action, so that data signals are directed toward the DEMOD and voice signals are directed toward the AGC position amplifier.

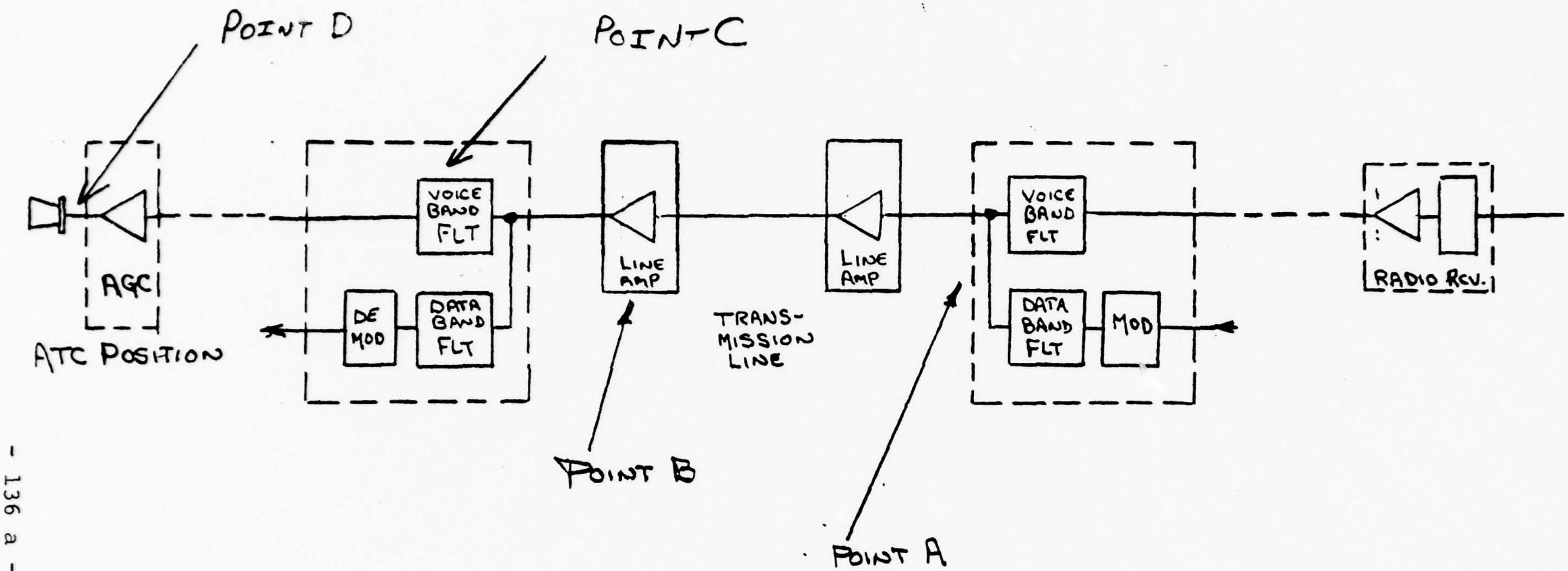


Figure 2.12-1

TYPICAL VOICE PLUS IN-BAND DATA CIRCUIT ARRANGEMENT (ONE-WAY ONLY) ILLUSTRATING AGC EFFECTS ON DATA SIGNAL REJECTION IN VOICE PORTION

Assuming that the filtering action on the data signal is such that there is a 70 dB relative attenuation of data carrier tone in the voice path toward the ATC position, the disturbing effect on the position listener is expected to be negligible due to the low level of interfering data tone. However, since the voice path includes an AGC amplifier, then the low level data signal will experience full AGC action. Hence, if the AGC amplifier can provide a 30 dB gain, then the expected low level interfering data tone signal will be raised by this amount and the relative 70 dB attenuation will become 40 dB. Due to this filter cancelling effect caused by the AGC amplifier, the interfering data tone now becomes a serious interference to the listener.

Therefore, if a voice plus in-band circuit arrangement is employed, either additional filtering must be provided or the system must be structured such that the AGC amplifiers are controlled. One control approach is to AGC the combined data tone and voice and locate the data removing notch filter after the AGC amplifier. In Figure 2.12-1, the voice band filter at point C would be moved into the ATC position to point D, which is after the AGC amplifier.

3.0 STUDY CONCLUSIONS

Based on the data obtained during the course of the study, several technical and operational conclusions can be derived. These are:

(1) With a slot bandwidth of 400 Hz, at the 3 dB points, in the voice band only slight degradation in speech quality (this is subjective) is observed.

With this slot bandwidth increase in the voice band, data is more efficiently transmitted since less voice spill over is encountered in the data bandpass.

(2) The location of the in-band data frequency slot is optimally within the frequency range of 2000 to 2350 Hz when using a transmission system of type 3002, unconditioned, and restricting the band between 2350 Hz and 2750 Hz due to the presence of telephone signalling tones at 2400 Hz and 2600 Hz.

If it can be determined that 2400 Hz is not used on FAA circuits, the data slot should be moved to the band between 2100 Hz and 2450 Hz.

(3) PSK modulation techniques offer the greatest benefits for developing a voice plus in-band data technique.

At a cost of increased error probability, but with decrease in circuit complexity in coherent detection, a differential phase shift keying (DPSK) may be considered. (26)

(4) Intelligible voice transmission and simultaneous 200 bit per second data, with error rates meeting the 1×10^{-4} error rate parameter specified, had been recorded as being technically attainable. Theoretical analysis indicates good voice quality can be achieved with another filter iteration and still meet the error rate requirement and data rates.

Additionally, it should be noted that error rates of 1×10^{-5} were attained during the laboratory tests under the conditions recorded in the test results, and with the limited number of 2 male and 3 female speakers.

(5) The 200 bit per second data rate is marginal for transmission of push-to-talk at the specified 80 ms round trip delay from activation of push-to-talk to return of confirmation.

Delay is significant during periods of normal operation and becomes worse if data errors are injected, or there is a loss of link character synchronization.

Multiple push-to-talk, as may be required for selective keying, appear to exceed the capability of the 200 bit per second channel.

(6) Status change orders should be encoded to ensure accuracy of data reception. It may not be possible to encode push-to-talk bits, due to the increase in delay time which results from encoding.

(7) T1 channel application is questionable.

(8) Additional research and development efforts are required to resolve those problems identified, and which can best be addressed using a test model of prototype quality rather than the current experimental model used for the laboratory tests.

As can be determined from the conclusions, the technique of inserting data in a voice band slot appears feasible, with qualifications, i.e., because of the number of parameters that cannot be assessed in the laboratory, or with the use of breadboard (experimental) type circuits, additional tests must be performed and circuit designs optimized. Furthermore, the approach must be analyzed with respect to specific FAA communication system application requirements. As indicated in paragraph 2.10 (coded/uncoded data), the 80 msec time constraint placed on push-to-talk/confirm data transmission in the slot is quite severe. The constraint becomes even more severe when multiple selective keying must be accommodated.

It is possible that certain less critical information (such as status reporting) may have to be deferred to allow more time for critical push-to-talk information. Also, a separate data tone for push-to-talk information is with consideration. The separate data tone would carry key active information with minimal delay, while a second data tone would carry "what to key" and status information. In this case, "what to key" information would be delayed only at the time of circuit selection but after selection actuations of push-to-talk would not be delayed.

4.0 RECOMMENDATIONS

Predicated upon the assumption that it is desired to further develop the technique analyzed under this contract, the following actions are recommended:

(1) Refine the design of the feasibility test model and implement the design employing prototype quality fabrication.

(a) Refinement in design would include level adjust controls, board layout and mechanical packaging.

(b) Optimize filter design to tenth order or possibly twelfth order for the band elimination filters. This is in lieu of the 8th order filter.

(2) Analyze the effects of AGC amplifiers in a test configuration and develop application recommendations.

(3) Analyze the effects on T1 channels using an actual set-up and develop application recommendations.

(4) If possible, define specific system applications so that system parameters can be defined and designs implemented to address this application in terms of a data slot technique.

Continuation of this analysis, using the labroatory feasibility model is not recommended, since under these conditions it would be, at best, extremely difficult to isolate and analyze those elements which are major contributors to performance degradation. This condition exists because of the material quality of feasibility models, which are, by their nature, constructed merely to prove or disprove the technical viability of a specific technique under study.

5.0 PRINCIPAL INVESTIGATORS

The study summarized in this document was conducted by ORION SYSTEMS INC. of Huntingdon Valley, Pennsylvania, as principal investigators.

In support of Orion Systems' technical staff, consultant expertise was provided by Professor Bruce Fritchman of Lehigh University and Dr. Peter Hahn. These technical experts provided the theoretical analysis of the configuration under test, which confirmed empirical test results to be within the technical bounds of theory.

In addition, Mr. George Warren, Engineer, of National Semiconductor, provided design parameters for the construction of the filters. These parameters were obtained from computer readouts of the filter design algorithm which National Semiconductor uses in their design of filter devices.

And, finally, the FSK test model (Ref. Appendix B) was designed and furnished by ARK Electronic Products, Inc., in Melbourne, Florida, who is an established vendor of Frequency Division Multiplex Equipments used for voice and/or data transmission.

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APPENDIX "A"

TEST PLAN

FOR

240 Hz SLOT STUDY PROGRAM

(CONTRACT NO. DOT-FA78WA-4099)

PREPARED BY

ORION SYSTEMS INC.

February 1978

Rev. A

TEST PLAN

240 Hz SLOT STUDY PROGRAM

1.0 GENERAL

1.1 OBJECTIVE - The object of these test procedures is to determine the effect in speech transmission and bit data rate capability in a 240 Hz frequency slot in the voice band, with .9999 efficiency. The test circuit shall use a Phase Shift Key (PSK) modulation technique at a carrier frequency of 2210 Hz. Tests shall include the simultaneous transmission and reception of voice and data to determine the effects each has on the other. In addition, noise signals will be injected to simulate telephone line interferences. An artificial transmission line (Axel Model 771) will be used to simulate the specified telephone line, namely; a 3002 unconditioned line.

The first two sections of the procedure "Speech Evaluation and Data Rate Transmission" were prepared to set the limits for each of the two signals (voice and data) separately. The third section then combines the two signals, utilizing the results of the first two sections, to determine the efficiency of transmitting and receiving voice and data in a voice frequency spectrum which utilizes a notch in the voice band for the transmission of data.

1.2 TEST EQUIPMENT

a. Digital Pattern Analyzer - Hewlett Packard Model 1645A (1 required).

b. Master Voice Tape, Male and Female Speakers, full voice bandwidth with sentences and phonetically balanced (PB) words.

c. Voice Tape Recorder - Sony Model TC-105A (2 required).

d. Artifical Transmission Line - Axel Model 771 (1 required).

e. Power Supply (± 12VDC) - Hewlett Packard Model 6205B (1 required).

f. Oscillosocope - Tektronix Model 465 (1 required).

NOTE: Equivalent Test Equipment models are acceptable.

2.0 TEST PROCEDURE

2.1 SPEECH EVALUATION TEST

2.1.1 Connect the test circuit to the test equipment as shown in Figure 1.

2.1.2 Set the artifical transmission line for a 3002 unconditioned line. All other line controls should be "off" or "out" of circuit.

2.1.3 Turn on test circuit power supply.

2.1.4 Turn on tape recorders #1 and #2.

2.1.5 Utilizing pre-recorded voice tapes, record a minimum of 30 seconds of a male voice transmission.

2.1.6 Turn off tape recorders.

2.1.7 Repeat the test in its entirety for a female speaker.

2.1.8 At the completion of this test, perform a subjective evaluation of the effects that the notch filters have on the speech quality.

NOTE: It may be necessary to repeat these tests with more male and female speakers to obtain a valid evaluation.

2.2 DATA RATE TRANSMISSION TEST

2.2.1 With the test circuit connected as shown in Figure 1, set the pattern generator for a 4 bit pattern.

2.2.2 Set the artifical line for a 3002 unconditioned line. (All other line controls should be "off" or "out" of the circuit.)

2.2.3 Transmit the 4 bit pattern at 150 bits/second rate. (Continuously transmit the pattern.)

2.2.4 No errors should be observed on the bit error rate counter.

2.2.5 Increase the artifical transmission line output attenuator until errors are observed. Record the attenuator setting and bit error rate. If no errors are observed, set the attenuator to -26dBm. If errors are observed, set the attenuator to a setting where no errors are observed for a minimum of 5 minutes. Maintain this setting.

2.2.6 Turn on the random noise generator of the artifical transmission line test set.

2.2.7 Set the random noise generator for a minimum output, i.e., maximum attenuation of the random noise generator.

2.2.8 Decrease the noise generator attenuation until one error in 10,000 bits transmitted is observed on the bit error rate counter. Test should run for approximately 67 seconds. (10,000 bits $\div$ 150 bits/seconds $\cong$ 67 seconds.)

2.2.9 Increase the noise generator attenuation until no errors are observed for a period of 5 minutes and maintain this setting.

2.2.10 Record the signal to noise ratio.

2.2.11 Turn on the impulse noise generat[illegible] of the artifical transmission line.

2.2.12 Set the impulse noise generator for a minimum output, i.e., maximum attenuation of the impulse noise generator.

2.2.13 Set the impulse rate to .2 pps.

2.2.14 Decrease the impulse noise generator attenuation until one error in 10,000 bits transmitted is observed on the bit error rate counter. Test should run for approximately 67 seconds. (If one error in 10,000 bits cannot be obtained, manually insert an impulse signal at 1 bit/minute and observe its effect on the BER counter and do not continue tests.)

2.2.15 Increase the noise impulses - control located on artifical transmission line - until 1 error in 1000 is observed.

2.2.16 Increase the impulse noise generator attenuation until a maximum of one error in 10,000 bits transmitted is observed on the bit error rate counter.

2.2.17 Record the settings of both the impulses/second and the attenuator.

2.2.18 Repeat procedures 2.2.14 thru 2.2.17 until the impulse noise transmission will not allow desired signal transmission of one bit error in 10,000 bits.

2.2.19 Repeat the entire "Data Rate Transmission Test" (procedures 2.2.1 thru 2.2.18) for bit rates of 200, 300, and 600 bits/second.

2.2.20 Repeat the entire "Data Rate Transmission Test" (procedures 2.2.1 thru 2.2.19) for various patterns as available on the Digital Pattern Analyzer, e.g., 3:1, 7:1, and 63 bits.

2.3 SPEECH PLUS DATA TRANSMISSION TEST

2.3.1 Connect the test circuit to the test equipment as shown in Figure 1.

2.3.2 Utilizing the test results recorded in section 2.2, set the pattern generator and artifical line to the control settings that allow for maximum transmission efficiency; i.e., minimum error rates when the output level of the line is at a minimum, the random and impulse noise signals are at a maximum, and the pattern and bit rates are also at a maximum.

2.3.3 Turn on both tape recorders, the pattern generator, and the BER counter.

2.3.4 Utilizing the pre-recorded voice tape, record 1 minute of a male voice transmission and record the BER on the bit error rate counter.

2.3.5 Set the voice input to the artifical transmission line to be 5 dB lower than the modulation tone.

2.3.6 Record 1 minute of the male voice, and record the BER on the bit error rate counter.

2.3.7 Set the voice input to the artifical transmission line to be 5 dB higher than the modulation tone.

2.3.8 Record 1 minute of the male voice, and record the BER on the bit error rate counter.

2.3.9 Repeat steps 2.3.4 thru 2.3.8 for a female voice.

2.3.10 With the voice and modulation tones set for the same output, turn on both tape recorders, the pattern generator and the BER counter.

2.3.11 Utilizing the pre-recorded voice tapes, record 1 minute of a male voice transmission and record the BER on the bit error rate counter.

2.3.12 Repeat steps 2.3.10 thru 2.3.11 for a female speaker.

2.3.13 Evaluate the voice tape recordings for the effect of the modulation tones on the speech signals.

2.3.14 Evaluate the bit error rate data for the effects that the voice signals have on the transmission of the data in the notch. If the error rates are intolerable, the bit rate, noise injection signals and attenuation parameters may require changes to improve the data transmission efficiency. That is, the word size may need to be reduced, the bit rate increased or decreased, and the noise signals decreased in order to increase the signal to noise ratio. These parameter changes should all be recorded and test continued until it has been determined, conclusively, if it is possible to efficiently transmit and receive data in a frequency slot of 240 Hz within the voice band without degradation to the voice or data signal.

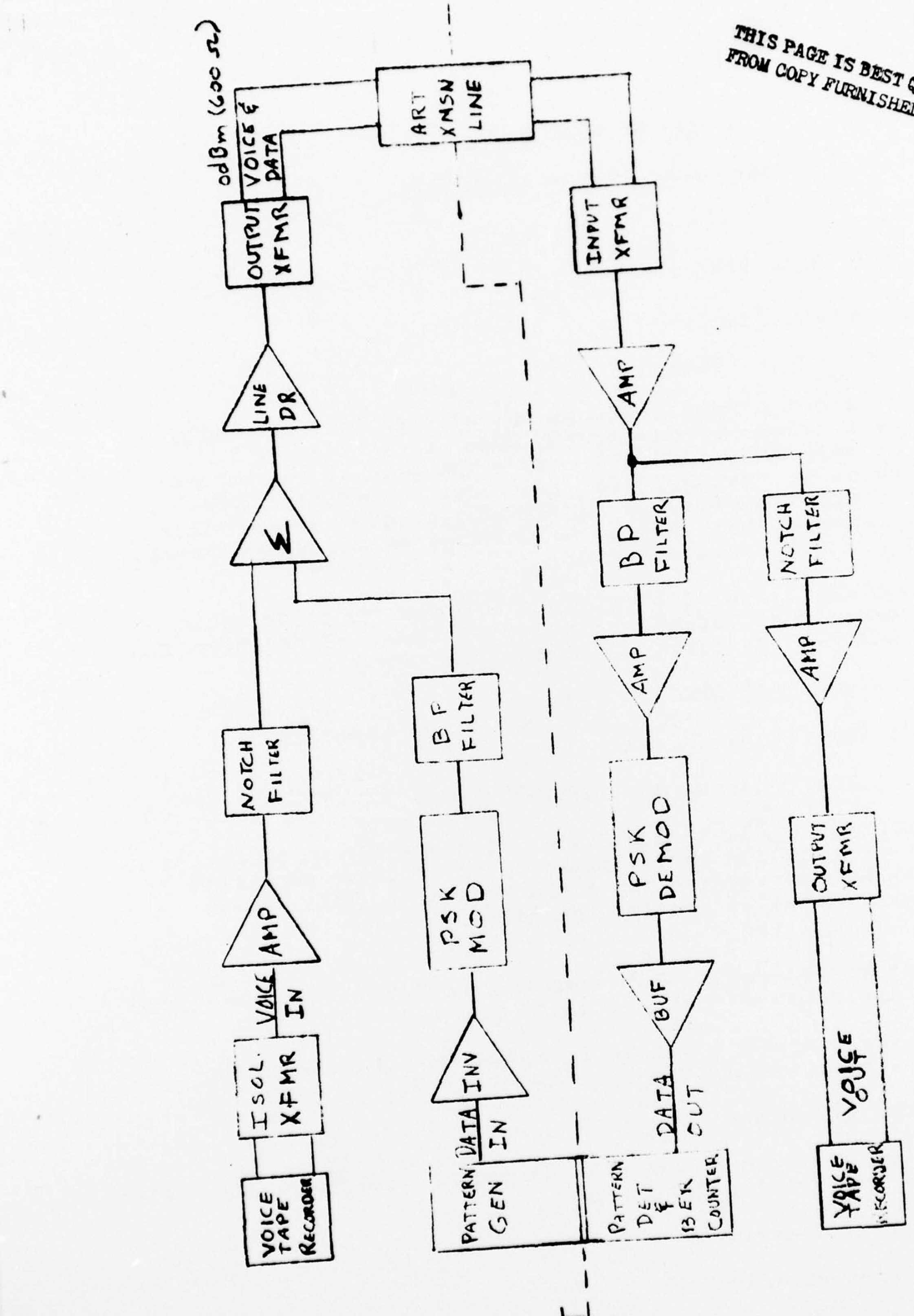

FIGURE 1
TEST SET UP

240 Hz Slot Study

Data Rate Transmission Test

Results (Ref. Test Plan - Para. 2.2)

A. 150 bps (Para. 2.2.1)

1. 4 Bit Pattern (0001)

2. Para. 2.2.5 - This paragraph was altered for this test, since in order to recover the data at the receive end, it is only necessary to increase the gain of the receive circuits to compensate for line loss. Therefore, the artificial line was set for minimum attenuation and data was recorded with respect to readings on the AC voltmeter, e.g., signal to noise ratio.

3. Paragraphs 2.2.8, 2.2.9, 2.2.10 - Signal to noise ratio as measured at output of artificial line. (9 dB attenuation on switches for noise attenuation setting.)

   Data Signal = 0 dBm)
) No errors observed.
   Noise (Random) = -20 dBm)

   Note: Data input to PSK DEMOD $\cong$ 15V pp.

4. Paragraphs 2.2.11 thru 2.2.15 - 10 pps at 70 mv peak to peak were applied at the input to the receive amplifier (secondary of receive input transformer). This was the maximum output of the impulse noise generator on the artificial line.

   No errors observed. This test was performed with the signal to noise ratio as recorded in A3 above.

5. With the settings as above in A3 and A4, the frequency translation setting was set for (-) 10 Hz.

   No errors observed.

6. With the settings as above in A3 and A4, the frequency translation was set for (+) 10 Hz.

   No errors observed.

7. 8 Bit Pattern (00000001)

   Steps 2 thru 6 inclusive (above) were performed with the 8 bit pattern and the same results were obtained, i.e., no errors observed under the test conditions noted.

8. 63 Bit Pattern (pseudo random)

Steps 2 thru 6 inclusive (above) were performed with the 63 bit pseudo random pattern and the same results were obtained, i.e., no errors observed under the test conditions noted.

B. <u>200 bps</u>

1. 4 bit pattern (0001)

2. 8 bit pattern (00000001)

3. 63 bit pseudo random pattern

With conditions as stated above in Section A, all tests were repeated for 200 bps and the same results obtained, i.e., no errors observed.

C. <u>300 bps and 600 bps</u>

Data measurements at 300 bps and 600 bps indicated unacceptable performance. Technical rationale for this condition focuses upon the filter characteristics and slot bandwidth as the first order of difficulties. Hence, in order to attempt data rate transmissions in the 300 to 600 bps range, far more stringent requirements must be imposed on the filter design and a redefinition of the slot bandwidth would be required.

240 Hz Slot Study

Speech Plus Data Transmission Test

Results (Ref. Test Plan - Para. 2.3)

Para. 2.3.1 - Under this test set-up it was observed that unwanted voice frequencies received by the bandpass filter (receive section) were passed on to the PSK Demodulator circuits and appeared as data signals, causing data errors. To compensate for the attenuation, the voice level was lowered and the following results were recorded.

A. Transmit Output (To Artifical Trans. Line)

1. Data Level (pseudo random 63 bit pattern @ 200 bps)

   0 dBm

2. Voice Level - 10 VU max. on voice peaks as measured with a VU meter across the transmit output transformer. (PSK modulator disconnected.)

B. Receive Input (From Artifical Trans. Line)

1. Data Level = -16 dBm

2. Noise Level = -36 dBm (attenuation switch set to 20 dB)

3. Voice Level = 16 dB below transmit level (VU meter could not record).

4. Impulse Noise set to 10 pps and 70 mvpp recorded at receive input transformer secondary.

5. Artifical Line set for +10 Hz frequency translation.

No errors were observed with the above conditions.

The test was run again with the same conditions as stated above, except the frequency translation was set for -10 Hz. Again, no errors were observed.

The voice tape used for the tests contained sentences and PB word groups spoken by 2 male and 3 female speakers, and the output of the voice circuits were monitored to be sure that the amplifiers in the voice circuits provided enough output to ensure that the voice could be recovered. (No AGC amplifiers were used.)

C. A voice plus data test was performed and a tape made with no errors observed, which is included as part of this test. Because of the noise generation by the experimental circuits, the noise generated by the artifical line was not used; only the frequency translation (+10 Hz) of the artifical line was included. As can be heard on the experimental tape, the voice was audible and the words spoken could be understood. However, the data signal could be heard in addition to the noise generated by the experimental circuits. As the theory in paragraph 2.5.1 of the final report indicates, an improvement in voice level could be provided, and the data signal spill over in the voice circuits could be reduced with more attenuation of voice frequencies in the data signal range. This would also allow for higher voice signals to be transmitted since, conversely, no voice spill over would be introduced into the data recovery circuits.

The following paragraphs cover the words and sentences used for the speech evaluation tapes.

"This tape was made to evaluate the effect of notching out a selected band of frequencies within the normal voice bandwidth. The total bandwidth of the notch, that is, the transmit and receive circuits in series, is 403 Hz at the 3 dB points, 310 Hz at the 30 dB points and 291 Hz at the 40 dB points.

The words selected for the tape are Phonetically Balanced (PB) groups of isolated words taken from a FAA Report, Number FAA-RD-77-153, and are spoken by 2 male and 3 female speakers. Additionally, an artifical transmission line, simulating an unconditioned 3002 line, is used as a transmission medium between the transmit and receive sections.

P.B. Words

STAB	CHUNK	TROUP	CLOTHES	PENT
TUCK	LATCH	CHEST	ROOMS	THOUGH
DRAPE	CODE	BOOST	LAG	JUG
PITCH	LOW	CLOWN	THIGH	SNIFF
INK	SHOT	DITCH	WAIT	QUIZ "

APPENDIX "B"

ALTERNATE TECHNIQUE TEST CONFIGURATION

In addition to the test measurements described in Appendix "A," an alternate test configuration was constructed and measurements made for comparison to the test model approach.

The alternate approach employed frequency shift keying (FSK) techniques and a passive filter for the data band elimination section. The filter associated with the standard FSK XMTR/RCVR units was used for the data passband filter section. These units were off-the-shelf FSK modules providing Bell or CCITT compatible data channel frequencies. The data channel employed was the CC4, with a center frequency of 2040 Hz; and a frequency deviation of ± 120 Hertz.

The characteristics of the data band elimination section were as follows:

Filter Section	Bandwidth (3 dB)
Data Band Elimination XMT Section	874 Hz (2480 to 1606 Hz)
Data Band Elimination Rec Section	834 Hz (2478 to 1644 Hz)

Test results indicated a degraded voice quality compared to the Third Iteration Test Configuration with PSK techniques. Data transmission was observed to be comparable.