Telesat

Telesat Canada

STUDY OF THE MSAT UHF GROUND TERMINALS

TASK 22

PREPARED FOR

DEPARTMENT OF COMMUNICATIONS

Submitted By: Telesat Canada

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TABLE OF CONTENTS

CHAPTER I - INTRODUCTION

TABLE OF CONTENTS (CONTINUED)

CHAPTER III - DATA TRANSMISSION APPLICATION TO MSAT

TABLE OF CONTENTS (CONTINUED)

CHAPTER IV - SPECIFICATIONS FOR MSAT UHF TERMINALS

TABLE OF CONTENTS (CONTINUED)

TABLE OF CONTENTS (CONTINUED)

CHAPTER V - COMPATIBILITY OF MSAT WITH TERRESTRIAL SYSTEMS

CHAPTER VI - THE MSAT L-BAND GROUND TERMINAL

TABLE OF CONTENTS (CONTINUED)

CHAPTER VII - CONCLUSION AND DISCUSSIONS

APPENDIX A - ACSSB TERMINAL REQUIREMENTS

TABLE OF CONTENTS (CONTINUED)

I - INTRODUCTION

1.1 Purpose of The Study

MSAT is a proposed geostationary satellite intended to provide voice and data communications services to mobile terminals operating in rural and remote areas of Canada.

In conjunction with this basic concept, Telesat was commissioned by the Department Of Communications (DOC) to study the requirements for mobile terminals operating in both digital and analogue modes for voice and data communications. While building upon the previous research work on the subject matter, the task at hand was to refine and update the specifications on the major parameters of the terminals, in line with the most current system design concept.

This report is intended to describe the outcome of these activities in the form of defining reasonable bounds for the terminal parameters principally based on the current understanding of the underlying technology.

1.2 Methodology

The major challenge facing the MSAT terminal designer is the fact that power and spectrum resources are in extremely short supply relative to the terrestrial mobile systems. The proposed frequency band of operation (821 MHz - 825 MHz, 866 MHz - 870 MHz) is in a very congested part of the spectrum and is only 8 MHz wide (Figure 1.1). Furthermore, due to the practical and economical implications of spacecraft hardware, there is only a limited frequency reuse capability foreseen for the early generations. In addition, the candidate terminal design concepts must conform to the characteristics of the mobile environment which exhibits extreme fading due to multipath and shadowing losses. In short, the terminal design must incorporate spectral and power efficient schemes, as well as hardware robustness and physical compactness, while maintaining an acceptable cost target.

To start the task, our approach was first to study the system constraints imposed by the link, as well as the objectives of voice quality for MSAT voice communications. During the MSAT Phase B Study, DOC awarded three major contracts to design and build MSAT terminals. SPAR Aerospace built an LPC/DMSK terminal; the LPC and DMSK portion being supplied by the DOC and Miller Communications Systems respectively. ADGA, with subcontracts to Canadian Marconi and J.P. McGeehan, formerly with Bath University, built an Amplitude Compandored Single Side Band (ACSSB) terminal. Glenayre Electronics Ltd., with a sub-contract to ADGA, built a Narrow Band FM (NBFM) terminal. Furthermore, a contract was awarded to BNR to conduct several tests in order to evaluate the subjective quality of speech of the ACSSB, LPC/DMSK and NBFM terminals, in which the ACSSB and LPC terminals were designed and built by CRC.

The current study started by reviewing the related reports generated during the MSAT phase A and B studies, including:

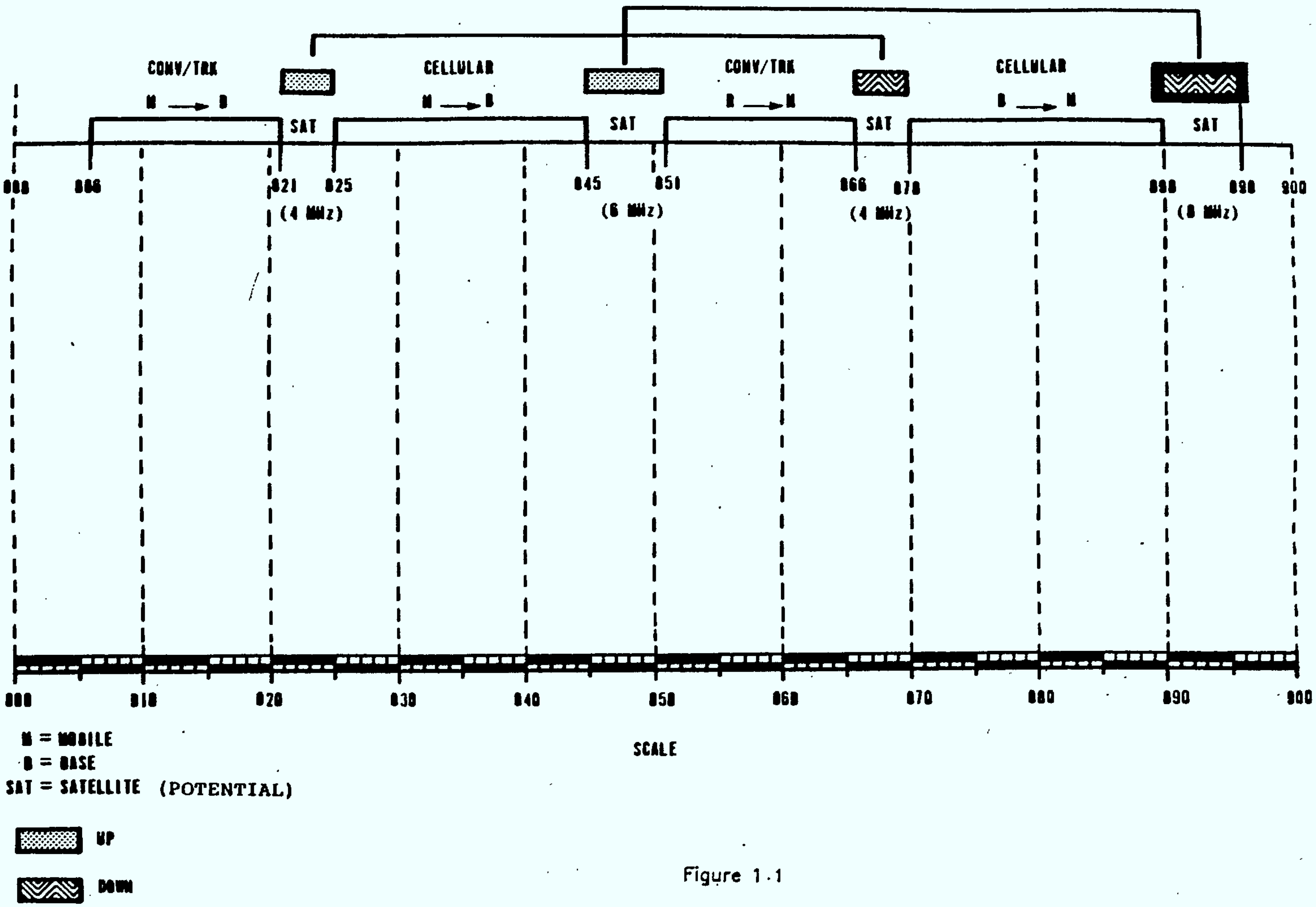

Figure 1.1

Current 800 MHz frequency allocation

- SPAR LPC/DMSK Terminal Design [1]
- SPAR Frequency Synthesizer Design [2]
- ADGA ACSSB Terminal Design [3]
- Woods Gordon Industry Impact Study [4]
- Telesat Commercial Viability Studies (CVS) and Business Proposal, [5] and [6].

However, the reviewing phase which was by no means limited to the studies mentioned above, covered a wide spectrum of relevant literature. The outcome of this extensive literature survey is reflected throughout the report. In particular, Chapter II provides a brief comparison of the four pilot tone configurations of ACSSB. It is shown that TTIB appears to be the most suitable technique for MSAT applications.

It is expected that data communications via MSAT will play a major role in MSAT services, especially in future generations. Since a small channel spacing is desirable, the requirement for a high performance, narrow band digital modulation is clear. Chapter III is devoted to data transmission and it provides a brief comparison of different modulation techniques. It appears that DMSK is the most favourable candidate to operate in the MSAT environment.

Chapter IV deals with the minimum performance requirements of an ACSSB terminal based on the current understanding of the relevant technologies. It is to be noted, however that the final performance requirements must be set in a fashion to meet the target price for terminal hardware, while maintaining an overall performance consistent with the baseline system design. This requires very close liaison with the equipment designers and hardware manufacturers. The dialogue must obviously be kept open until such time that the terminal design is mature enough for implementation.

Since the beginning of the MSAT concept, there have been suggestions that the design of MSAT terminals could be based on the existing terrestrial mobile radio designs. Thus, Chapter V is devoted to discussing the possibility of converting a conventional FM or ACSSB terrestrial radio to operate in an MSAT system. The views of several terminal manufacturers on this subject are also presented in this chapter.

Based on the results of the BNR subjective voice quality tests, conducted at UHF, and the current available fading statistics at L-band, the expected voice quality of ACSSB at L-band is estimated. Chapter VI addresses this issue and shows that the current Telesat L-band baseline design cannot provide the same mean opinion score as that of UHF for mobile terminals operating in fading shadowing environment.

Finally, in Chapter VII, the main conclusion of the work is presented with several recommendations in the areas requiring further R&D activities.

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2. Study of a Frequency Control System and Prototype Development of Frequency Sources and Synthesizers for the Mobile Satellite System Mobile Terminals. Vol.1-2, October 1983.

3. MSAT ACSSB Ground Terminal Study and Final Report by ADGA & Canadian Marconi. Vol. 1-5, DSS Contract No. 36001-2-2746-1, June 1985.

4. System Definition, Industry Capability Analysis, Export Market Analysis, Impact Analysis; by Woods Gordon. March 1985. DOC Contract No. OSM83-00001.

5. Telesat Commercial Viability Studies (CVS) Task 3, Systems Analysis. DOC Contract No. 01SM.36100-2-4372.

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II - MODULATION TECHNIQUES

2.1 Introduction

The proposed MSAT system, with its power and spectrum limitations, dictates the use of narrow-band technologies for voice and data communications.

For delivery of voice services, which basically may constitute the bulk of the identified demand for MSAT [2] in the first generation, Amplitude Compandored Single Sideband (ACSSB) and low bit rate Linear Predictive Coding/Differential Minimum Shift Keying (LPC/DMSK) are currently the only candidate schemes which could potentially meet the requirements demanded by MSAT spectrum and power resources limitations. These two options basically represent the system's analogue and digital alternatives, respectively. Despite its assumed better quality at high SNR and higher degree of hardware maturity, the conventional Narrow Band Frequency Modulation (NBFM) with 30 kHz channel spacing has been dropped from the list of modulation candidates for MSAT due to its excessive power and bandwidth requirements.

2.2 Proposed Techniques by U.S. Applicants to the FCC

In the United States of America twelve companies have filed to the Federal Communications Commission (FCC) to offer mobile satellite services (MSS). Their applications were reviewed to determine the modulation techniques that they are proposing for voice and data communications.

Table 1. FCC Filing Summary On Modulation Technique and Bandwidth

	Company	Modulation	Channel Spacing, kHz
1.	Hughes Communications Corporation	ACSSB	5
2.	Skylink Corporation	ACSSB, LPC for voice BRSM for data and control	5 5
3.	Omninet Corporation	ACSSB, LPC for voice GMSK for data	5
4.	Mobilesat Corporation	ACSSB	5
5.	MCCA American Satellite Services Corp.	NBFM for voice DPSK and QPSK for data	30
6.	McCaw Space Technologies, Inc.	ACSSB, LPC for voice GMSK for data	5
7.	Wismer & Becker/Transit Communications, Inc.	LPC for voice GMSK for data	5
8.	Mobile Satellite Services, Inc.	NBFM, LPC, ACSSB for voice PSK or FSK for data	30, 15, and 4
9.	N. American Mobile Satellite	ACSSB and LPC for voice	5
10.	Global Land Mobile Inc.	NBFM for voice M-ary PSK for data	15
11.	Satellite Mobile Telephone	ACSSB, LPC for voice	5
12.	Globesat Express	ACSSB for voice	5

2750Z/2751Z/2869Z

Table 1 summarizes the results. Note that all of the applicants, with the exceptions of MCCA and GLMS, have considered narrow band technologies for voice communications, and the proposed schemes are Amplitude Compandored Single Sideband (ACSSB) and digital transmission with Linear Predictive Coding (LPC).

2.3 BNR Voice Quality Test Results

In order to ensure the commercial viability of the MSAT system, it is important to select a cost-effective modulation scheme which also produces voice quality acceptable to potential users. In 1984, the Department Of Communications (DOC) commissioned Bell Northern Research Ltd. (BNR) to evaluate the subjective quality of three modulation/coding schemes, two of which were being considered by the Communication Research Centre (CRC) and Telesat as the candidate schemes for MSAT voice communications. The modulation techniques studied are:

- Narrow Band Frequency Modulation (NBFM)
- Amplitude Companded Single Sideband (ACSSB)
- Linear Predictive Coding/Differential Minimum Shift Keying (LPC/DMSK).

Note that NBFM was used only as a reference.

The salient results of the study are shown in Figures 2.1-2.3 [1]. Figure 2.1 gives the Mean Opinion Score (MOS) ratings for ideal conditions; i.e., no fading, no background degradation, and quiet listening environment. It is seen that ACSSB areas fares reasonably well with NBFM at high C/No ratios. But more significantly, it degrades much more gracefully than NBFM when C/No decreases; and it out-performs NBFM for C/No below 53 dB-Hz. For LPC, however, the rating is always very low regardless of the C/No value.

Figures 2.2 and 2.3 depict the performance of ACSSB and NBFM, respectively, under fading conditions using CRC's link simulator. The results for LPC are not given here since the rating has been always poor. Note that the light shadowing/fading statistics case is the one used in the Telesat baseline design. As it is seen in Figure 2.2, the ratings drop rapidly to poor quality for NBFM. However, Figure 2.3 shows that ACSSB degrades, again, gracefully in a mobile fading condition.

Note that the Telesat baseline design at UHF gives a C/No of about 50 dB-Hz which, for MRS users with ACSSB terminals, corresponds to a rating of 3 (fair) for light shadowing/fading environment.

The BNR voice quality tests, as well as the predicted advances in LPC/DMSK technologies in the next few years, suggest that, at least for the first MSAT generation, ACSSB should be the selected modulation scheme offered to the general public for voice communications. However, it is noteworthy that the subjects in the tests had been asked to evaluate the quality of the speech, rather than its intelligibility. Therefore, the so called "electronic accent" of LPC resulted in poor scoring by the subjects who were not familiar with this type of accent. In addition, the LPC algorithm used in the test was optimized for a male voice and, therefore, the MOS for

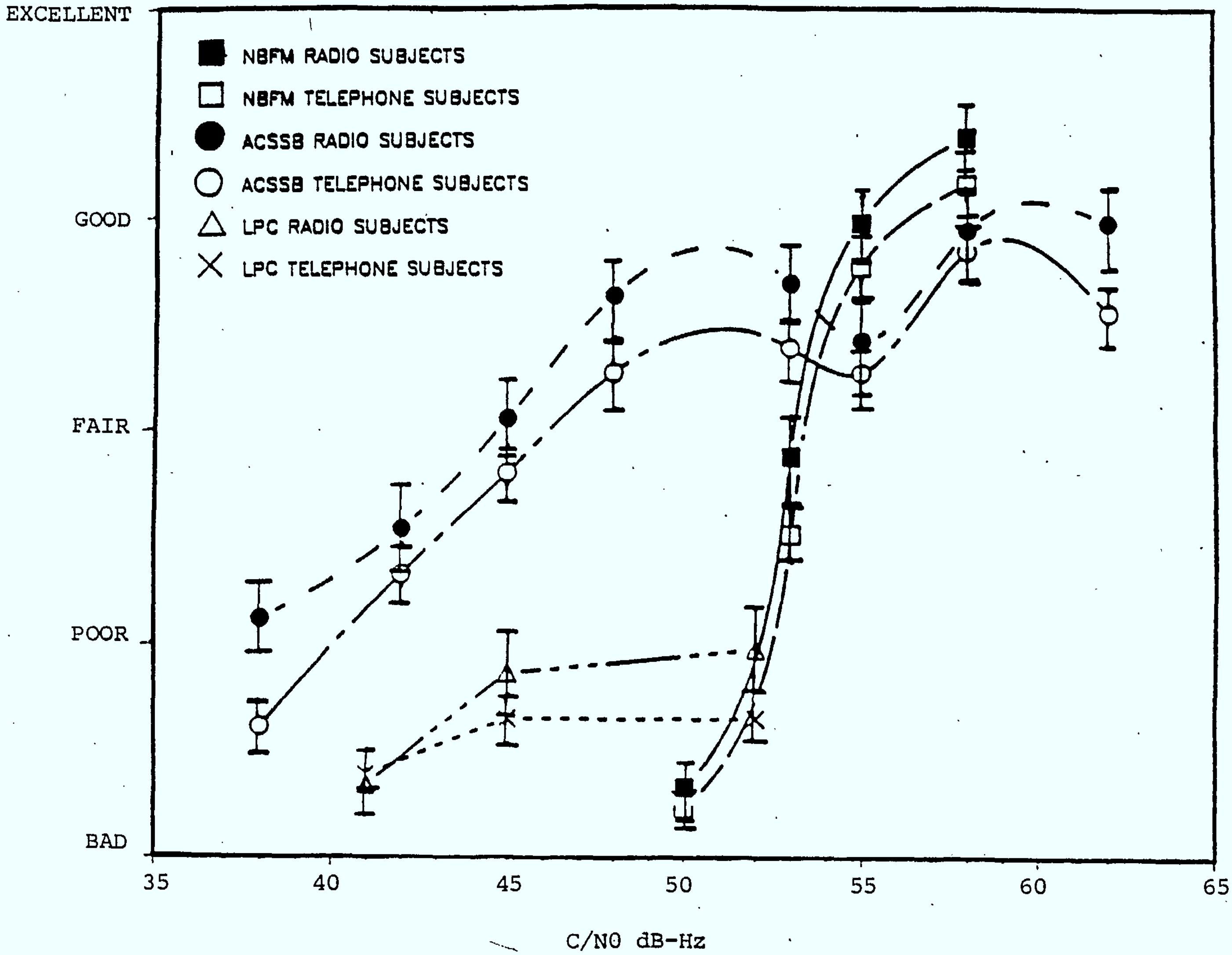

Figure 2.1- C/N0 vs Subjective Quality

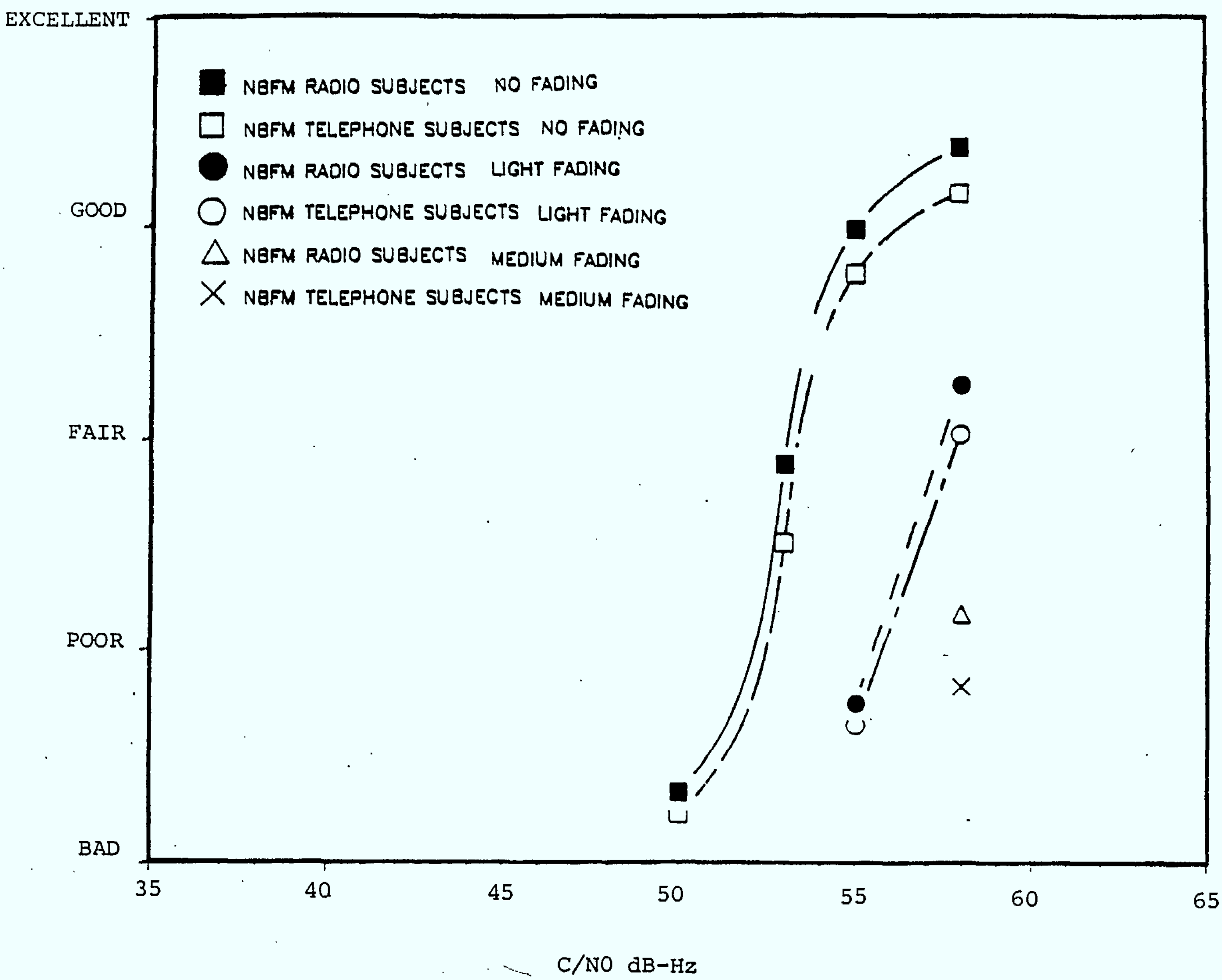

Figure 2.2- C/N0 vs Subjective Quality
for NBFM .

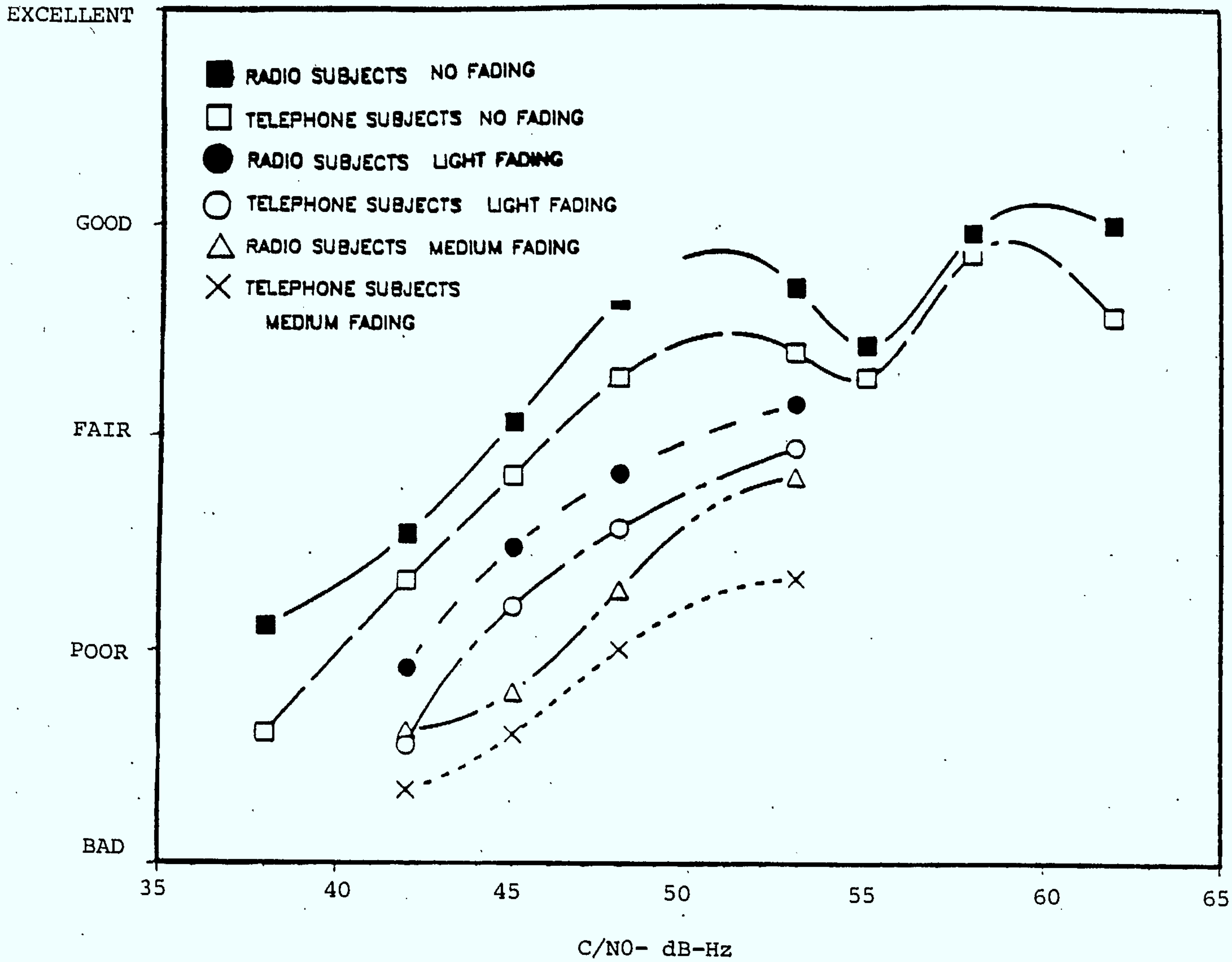

Figure 2.3- C/NO vs Subjective Quality
for ACSSB

male and female talkers were significantly different from each other. However, it is clear that LPC will be the choice of voice communications in those organizations where privacy of communication is of major concern, such as in military and law enforcement sections.

2.4 Review of the ACSSB Techniques

There are four single sideband configurations: [3]

- Tone-Below-Band (TBB)
- Tone-In-Band (TIB)
- Tone-Above-Band (TAB)
- Transparent-Tone-In-Band (TTIB)

The four configurations differ essentially in the positioning of the pilot reference, which is required for Automatic Frequency Control (AFC) and Automatic Gain Control (AGC) purposes in the receiver.

The advantages and disadvantages of each reference configuration are examined briefly in this section with the conclusion that the TTIB configuration is the most suitable configuration at UHF. For systems requiring analogue and digital transmissions, TTIB also seems to be the most favourable scheme. This is due to the fact that the pilot, which will be used for coherent detection in the receiver, is less exposed to phase and amplitude distortion of the IF filter, as will be discussed later.

a) Flexibility of the Transmission Channel:

TBB and TAB allow the use of the complete audio channel, typically 300 Hz to 3.4 kHz. With TIB, the removal of the central portion of the baseband spectrum to accommodate the pilot reference destroys the channel transparency to the majority of data formats as well as the ability to transmit speech at operating frequencies above 500 MHz. This restriction of TIB is overcome by using TTIB which does not require the removal of any portion of the spectral components and, thus, presents a transparent channel as with TBB and TAB configurations.

The TBB technique is restrictive in that sufficient spacing must be maintained between carrier reference and low frequency components in order to avoid interference as a result of Doppler spreading in the multipath environment. Also, the sub-audio signalling tones must not interfere with the AFC loop.

b) Transmitter Adjacent Channel Radiation:

With the narrow channel spacing envisaged for SSB mobile radio systems of 5 kHz, the spurious emission from the transmitter into an adjacent channel must be minimized. Non-linearity in the transmitter power amplifiers is the major factor in adjacent channel radiation producing intermodulation between reference and input modulation components. With TBB and TAB, where the reference is at the band extremes, the IM products fall into the adjacent channel. With TIB and TTIB, the majority of the intermod components fall within the designated channel thereby reducing the extent of adjacent channel radiation.

c) Receiver Processing Sensitivity to Adjacent Channel Interference:

The AGC and AFC circuits within the receiver are controlled by using the received reference which should be free from interference that is not correlated with the interference of the wanted modulation components. The susceptibility of the reference to adjacent channel interference should be minimized. TIB and TTIB, by virtue of the central reference position, are far less vulnerable to adjacent channel interference than TBB or TAB, and can also tolerate greater frequency errors in the adjacent channel transmitter frequency [3].

d) Reference Extraction:

The necessity for minimizing reference interference in the receiver control processing entails the use of high-order narrowband filters for reference extraction. In addition to the external interference, suppression of the transmitter modulation components in the neighbourhood of the pilot tone is required. Both TIB and TTIB are at a disadvantage with modulation components lying on either side of the reference. The problem is less severe for TTIB systems, since the spacing around the pilot or so-called "notch-width" can be increased at will within limits.

e) Transmit and Receive Processing Requirements:

With any reference configuration, the transmitter baseband (BB) filtering must be sufficient to ensure that the Doppler spread reference in the receiver can be extracted from the spread modulation components without ambiguity. The minimum reference-to-audio band separation in the transmitter, assuming ideal filtering in the receiver and highly stable local oscillators, is simply twice maximum Doppler frequency. However, the frequency stability requirement of ±0.1 ppm in the receiver increases the separation between the pilot and the audio band by ±90 Hz at 900 MHz. Thus, with an ideal filtering, the spacing between the two subbands in TTIB would be twice the sum of the doppler frequency shift and the frequency tolerance of the oscillator. This would be nearly 400 Hz at UHF. However, practical filtering would increase the necessary separation by a factor of approximately 1.5 to 2. That is, the required separation could be as much as 800 Hz at UHF, or 400 Hz from each subband. Note that 400 Hz spacing between the reference and modulation places a lower frequency bound on TBB, if the reference is to be placed at the carrier frequency, of 400 Hz, thus restricting the audio channel bandwidth. Similarly, with TIB, the 800 Hz notch required in the band center is unacceptable for quality speech and data communications.

Reference suppression at the receiver output is relatively simple for all four configurations, provided some form of multipath fading correction is employed to regenerate the reference as a single frequency component at the receiver output stages. Low-pass, high-pass and notch filtering are required for TBB, TAB and TIB, respectively, while reference suppression is inherent in the baseband processing of the TTIB system.

f) System Complexity:

The TTIB configuration is at a disadvantage, requiring several filtering and mixing stages, and phase-locking circuitry in the processing. There is a requirement for high-order filters for the three other techniques.

g) Correlation of Multipath Fading Between Reference and Wanted Modulation Components:

The audio processing technique needed in the ACSSB receiver to correct for amplitude and phase fluctuations is called Feed-Forward Signal Regeneration (FFSR) [4]. The FFSR technique removes any frequency error from the input signal, thus compensating for receiver mistuning arising from local oscillator drift in either the transmitter or receiver.

A high degree of correlation between fading of the reference and the wanted modulation components is essential for satisfactory performance of fast-acting FFSR. The major source of de-correlation in the receive processing is the IF selection filtering due to the non-linear phase characteristics and variable attenuation, particularly at the band edges. TBB and TAB are at severe disadvantages since the reference signal experiences different amplitude and group delay characteristics compared to those experienced by the majority of the modulation components in the central portion of the band [3]. Equalization of group delay and gain could be achieved for the reference. However, the local oscillator drift and temperature effects in the crystal filter would result in variable gain and phase characteristics and render this solution impractical. A solution to this problem is to provide a guard-band between the reference and filter edge such that the reference lies far from the nonlinear amplitude and group delay region of the IF filter. The drawback to this simple solution is the necessary increase in channel bandwidth to accommodate the guard band. With TIB and TTIB systems, the problem of fading decorrelation due to the IF filter is substantially reduced, since the reference lies in the linear zone of the filter, away from the band edge.

h) Channel Bandwidth:

Both TAB and TBB require a guard band which typically consists of a 400 Hz separation between the reference and filter band edge for MSAT UHF systems. Therefore, for an audio bandwidth of typically 3100 Hz, the minimum channel bandwidth for TAB and TBB is about 3.9 kHz. With TIB, the reference/component spacing is achieved by removal of a part of the audio spectrum, thus, only 3.1 kHz is required. TTIB systems require 3.9 kHz of bandwidth.

i) Frequency Acquisition Capabilities:

Due to local oscillator drift in both the transmitter and receiver, the demodulated SSB signal can experience significant frequency errors unless some form of feedback control system is used. The need for frequency control is especially present since highly stable oscillators are very expensive. The amount of frequency error which can be tolerated by the control system is governed by its acquisition range. The processing requires the reference component to be passed by the IF pilot filter, to generate the required error signal within the receiver. With TBB and TAB, where the reference is already at the band edge, it is possible, due to frequency errors, for the reference to fall outside the filter bandwidth. With TIB and TTIB, the range of frequency errors for which the reference remains within the IF bandwidth is up to one half of the filter bandwidth. This is sufficient to accommodate most oscillator drifts which will be encountered in the transceiver equipment.

j) Suitability for Data Transmission:

TIB, by virtue of its notched spectrum, is not transparent to conventional forms of data transmission, e.g. PSK, DPSK, FSK, MSK, etc. The three other systems provide a transparent channel, of which TTIB is particularly advantageous in that it can be used to achieve simple coherent data detection using the transmitted reference tone. Furthermore, this system configuration also facilitates a simple bit synchronization system at the receiver [3]. However, data transmission in an ACSSB channel is not expected to be a common mode of operation.

k) Receiver Demodulation Error:

As mentioned previously, feedforward frequency control can be used to reduce the frequency error but is dependent upon the frequency accuracy of the receiver reference. For TBB, the reference is at d.c. and thus accurate receiver demodulation with zero error is possible. With TAB, TIB and TTIB the required receiver reference is ideally the exact frequency of the transmitter reference. For TTIB, a simple modification of the circuitry allows the necessary reference information to be transmitted with the wanted modulation as a function of the sub-band separation, facilitating coherent demodulation of the SSB signal.

l) Intermodulation:

The intermodulation products of the modulated signal components and the tone would fall into the audio band if the pilot tone was centered in the channel, and into an adjacent channel if the tone was placed above or below the audio band. The IM products of the pilot tones in the case of TTIB will fall into the notch where the pilot is located in a way that, by filtering, they will be eliminated. In the TAB technique, IM products of the pilot tones will overlap on the audio signal.

From the above discussions, it appears that TAB and TTIB are the most suitable choices for analog speech transmission via MSAT. Although system complexity increases for TTIB, it could be overcome by the use of LSI techniques. In addition, by using the reference tone, TTIB seems to offer a simple coherent detection technique in an MSAT multipath fading environment.

2.5 TTIB Baseband Processing Requirements

The Transparent Tone-In-Band technique requires the following signal processing operations in the transmitter:

- Bandsplitting to separate the upper and lower sub-bands
- Automatic Level Control (ALC) to limit the speech level variations
- Compression
- Pre-emphasis
- Pilot insertion

The receiver baseband processing requirements are:

- Feed-Forward Signal Regeneration
- Separation of the upper sub-band, lower sub-band, and pilots (as described below)
- Expansion
- De-emphasis
- Sub-band Combining

The following briefly addresses these steps.

2.5.1 Bandsplitting

The bandsplitting operation is done using two mixers at frequencies f_1 and f_2 where $f_2 - f_1$ is the notch width. By varying f_2 and f_1 the notch width could be increased at will. The baseband signal, which is usually accepted to extend from 300 Hz to 3 kHz, is split into two approximately equal frequency segments, e.g. 300 Hz to 1.7 kHz and 1.7 kHz to 3 kHz, by suitable filtering. The upper frequency band is translated up in frequencies by an amount equal to the required notch width and then added to the lower frequency band. Figure 2.4 illustrates the principle of the bandsplitting operation.

2.5.2 Automatic Level Control (ALC)

The purpose of performing the ALC operation is to attenuate all audio signal peaks exceeding a pre-determined amplitude so that the modulated signal will similarly show no high peaks before being input to the high power amplifier. This will prevent the power amplifier from being overdriven and, thus, the IM distortion will be minimized. In addition, the speech will have a constant average power after being processed in the ALC circuit. The effects of the ALC circuit can be summarized as:

- Limiting the variations in the speech
- Minimizing the distortions created by the compandor since the variation in the compressor gain will occur at a slow rate. This will enable the expander in the receiver to follow closely the speech waveform processed by the compressor.
- Having less variations in the speech will facilitate the operation of the AGC circuit which corrects for fading.
- By limiting the speech power, all users will be sharing the satellite power equally.

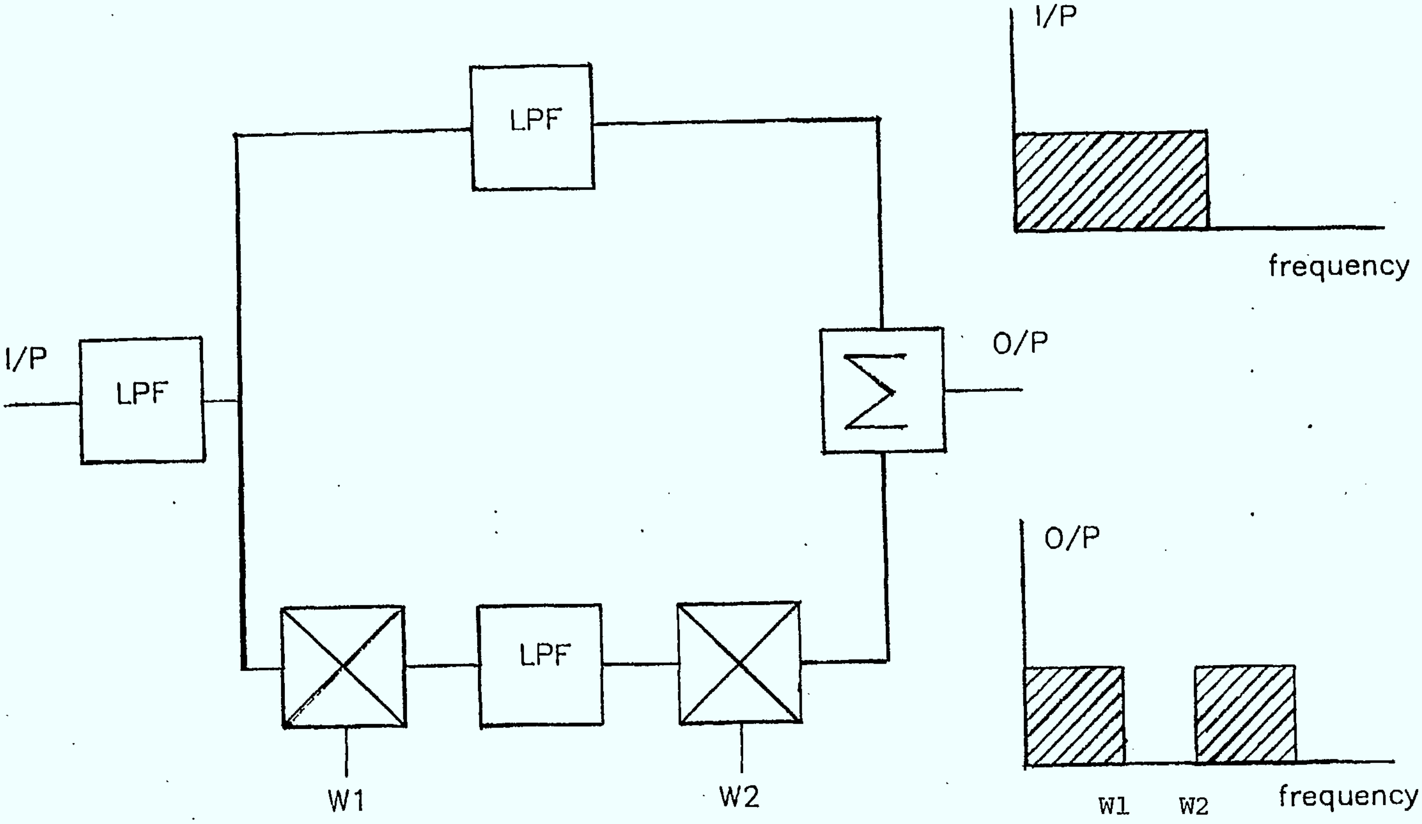

Figure 2.4 : Bandsplitting operation

2.5.3 Pre-Emphasis

The pre-emphasis circuit is a high pass filter that will attenuate the low frequencies and boost the high frequencies. The spectral shape of an unprocessed voice signal shows a marked difference in power between the low frequency peaks and the high frequency components. Therefore, we cannot expect a complete flattening of the spectrum with the use of pre-emphasis.

The use of pre-emphasis on SSB speech transmission should show some improvement in terms of signal-to-noise ratio (S/N). That is, in the presence of noise, one can expect a better subjective S/N at the output of the receiver.

2.5.4 Pilots Insertion

In the TTIB configuration, a pilot is added at the centre of the notch created by splitting the audio spectrum. In the technique developed at CRC, one main pilot and two additional subpilots are inserted in the notch. The main pilot will be used in the receiver as a reference to correct for amplitude and phase fluctuations occurring on the composite signal on the path between the transmitter and the receiver. The sub-pilots could be used in the receiver to recombine the upper and lower sub-bands. The main pilot power level, which depends on the minimum channel bandwidth in TTIB, will vary proportionally with it in order to maintain the required C/No ratio. The pilot IF filter should be able to separate the main pilot from the composite signal without distorting the sub-pilots.

For power saving purposes, the system could be designed so that the pilot would go on and off when the speech is present or absent, respectively. (Voice activated pilot)

2.5.5 Feed-Forward Signal Regeneration (FFSR)

Single Sideband, being an AM-type modulation, is subject to both the random envelope and phase induced fluctuations due to a multipath environment. FFSR is required to suppress these unwanted variations from the receiver output. The FFSR technique also removes any frequency error from the input signal, thus compensating for the receiver mistuning arising from local oscillator drift in either the transmitter or the receiver. A high degree of correlation between fading of the pilot and wanted modulation components is essential for satisfactory performance of FFSR, i.e. the pilot fading must be representative of the channel fading.

To illustrate the FFSR operation, a simple mathematical description will follow. Let y(t) be the received faded signal [4].

$$y(t) = r(t) \left[A \cos ((\omega_1 + \omega_p)t + \theta(t)) + B \cos ((\omega_1 + \omega_s)t + \theta(t))\right]$$

where $r(t)$ and $\theta(t)$ represent the amplitude and phase fluctuations respectively.

ω_1 = audio IF frequency of the receiver
ω_s = audio signal frequency components
ω_p = pilot frequency
A, B = amplitudes of the pilot and audio signal, respectively

The action of FFSR is to generate a signal n(t) at a frequency ω_o given by:

$$n(t) = C/r(t) \cos (\omega_o t + \theta(t))$$

where C is a constant. By mixing y(t) with n(t), the resultant signal will be:

$$\begin{aligned} Z(t) &= AC/2 \cos (\omega_p t + (\omega_1 - \omega_o)t) \\ &\quad + BC/2 \cos (\omega_s t + (\omega_1 - \omega_o)t) \end{aligned}$$

If the receiver is configured such that ω_1 equals ω_o in the above expression, then the required signal is coherently demodulated to baseband. The resultant signal, Z(t), is free from random amplitude and phase modulations.

2.5.6 De-Emphasis

This is a noise-suppression technique which can be integrated into the ACSSB design to improve the subjective quality of the speech signal.

The de-emphasis circuit shapes the noise floor in such a way that the total noise power in a given (f_2-f_1) bandwidth, where f_1 and f_2 are the lower and upper frequencies, respectively, is reduced by approximately: $10 \log (f_2/f_1)$. Pre-emphasis is effective only in reducing the high frequency noise components (see Figure 2.5). For example, a 6dB/octave receiver deemphasis, when used with corresponding transmitter pre-emphasis, achieves a noise energy reduction of 3.2 dB, while with a 12 dB/octave de-emphasis, the improvement is increased to 4.3dB [3].

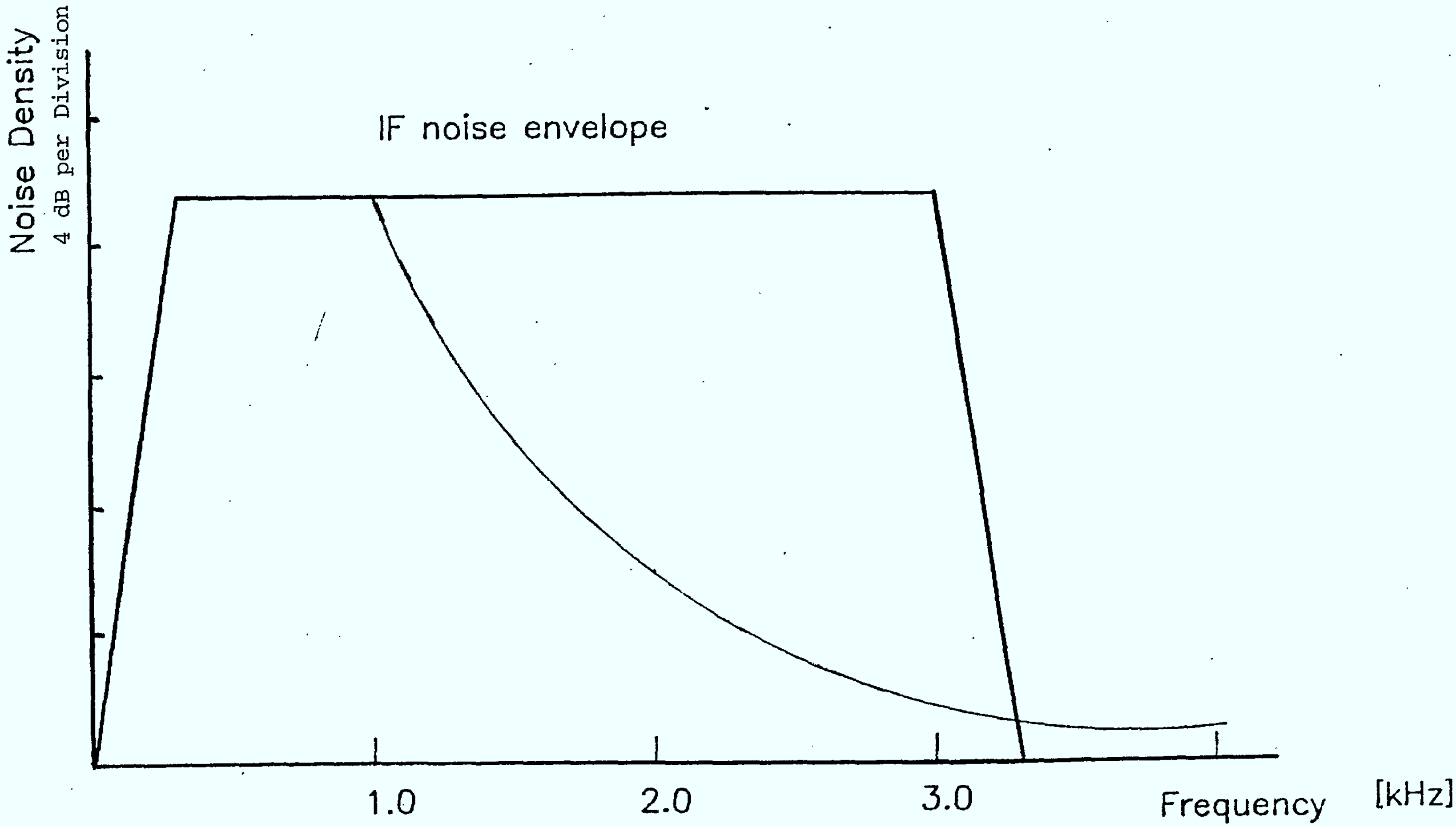

Figure 2.5 : Noise suppression due to De-emphasis
12 dB/octave @ 1 kHz

2.5.7 Sub-Band Combining

For the CRC technique, the sub-pilots are used for sub-band combining. The upper sub-pilot is associated with the lower sub-band and vice versa. The final stages of the audio processing in the receiver remove the pilot in the usual way for AGC and AFC purposes and perform a complementary downward frequency translation of the upper half of the audio spectrum, thereby regenerating the origin baseband signal.

2.5.8 Companding

The inclusion of compandors in SSB would improve the subjective signal-to-noise ratio (S/N) [10]. The purpose of the compandor in a speech communication link is to reduce the effect of noise on the received signal quality. The basic idea consists of compressing, i.e. reducing, the dynamic range of the speech signal on the transmitter side and expanding the signal on the receive side. The overall effect of companding is to greatly decrease the receiver output link noise when speech is absent and mask the noise when speech is present. The compressor and expander are two non-linear devices with complementary characteristics. The channel bandwidth should include all the components produced by the compressor, otherwise the restored signal would be distorted. The amount of distortion produced by compandors is very little, and since this distortion is harmonically related to speech, it is quite imperceptible.

One of the difficulties in determining the compandor noise improvement and the compandor loading is that the gains of the compressor and expander vary with the magnitude of the input signals. That is, the improvement in signal-to-noise ratio depends on the input signal level, and the lower the signal level, the less the companding advantage that can be achieved by a compandor.

2.5.8.1 Attack and Recovery Times of a Compandor

Compression is not instantaneous but occurs at a syllabic rate in which the compressor's gain varies with the envelope of the speech signal. Two time constants are required to resolve the signal power: the attack and recovery times. The attack time is the time needed by the compressor to set its gain or attenuation. If the attack time is too short, a very wide bandwidth will be required to transmit the signal, otherwise the expandor will be unable to accurately reconstruct the signal. In the absence of speech, the noise sets the compressor's amplification, and if the signal is suddenly present, the gain will be initially too high. In other words, overload will occur if the attack time is too long.

The recovery time is the period necessary for the compressor to adjust to a lower power level once a previous gain has been set. If the gain does not change quickly enough then the noise between short pauses will be amplified and may become unacceptable.

2.5.8.2 Compandor Noise Mechanisms

Compandors function by using speech to mask noise when speech is present and by reducing the noise when the speech is absent. This is done by varying the gain of the expandor according to the level of the speech or noise at the input to the expandor. To maximize the subjective noise reduction of the compandor, it is critical that the expandor follows the speech waveform as closely as possible. The compressor acts as the complement to the expandor, both in controlling its gain and in the response times associated with the gain changes. If the compressor and expandor do not have complementary responses, then an end-to-end distortion will occur. Thus, the accuracy with which the expandor follows the waveform must be matched by the compressor. In the case where the attack and recovery times are long, a burst of noise will be heard at the end of each speech syllable, while the gain of the expandor is being reduced.

Three types of noise can be identified in a companding system:

1. The noise heard at the end of the syllable
2. The noise heard when the speech is present
3. The noise heard with the absence of speech.

Compression ratios greater than 2:1 can be used but generally are not for the reason that higher compression ratios result in a greater requirement in the gain stability of the path [6].

2.5.8.3 Instantaneous Channel Power Before and After Compression [7]

a) Talker Power, $P_1(x)$

The telephone talker power is normally distributed with a mean of -23 dBmO and a standard deviation of 4.8 dB.

$$P_1(x) = 0.083 \text{ Exp } (-(x + 23)^2/46.08) \qquad (1)$$

where x is the instantaneous talker power.

b) Speech Amplitude Density Function, $P_2(r)$

The speech voltage is best characterized by a Gamma density function, $\Gamma(m)$, as

$$P_2(r) = (K^m \, r^{m-1} \, e^{-Kr})/\Gamma(m) \qquad (2)$$

where m and K are constant parameters. With m = 0.2, $K = \sqrt{m(m + 1)} = 0.4899$, r is the instantaneous voltage, and $\Gamma(m) = 4.591$

c) Instantaneous Power in 40% Active Channel

The instantaneous power P_3 (xi) in a 40% active channel is:

$$p_3(xi) = 0.4 \sum_{xi-20}^{3.8} P_2 \; (xi - x) \; P_1(x) \qquad (3)$$

where $P_2(xi - x) = 0.115 \, rp_2(r)$

xi is the instantaneous power,

and P_1 and P_2 are the functions defined in a and b above.

Substituting P_1 and P_2 by the expressions in equations 1 and 2, and replacing r by $10^{(xi-x)/20}$ in equation 3, one can get:

$$p_3(xi) = 6.8 \times 10^{-4} \cdot \Sigma\ 10^{0.2(xi - x)/20} \bullet e^{-0.49 \bullet 10^{(xi - x)/20} + (x + 23)^2/46.08}$$

The cumulative function of the instantaneous power is:
$P_3(xi) = xi\ \Sigma\ p_3(xi)$

The probability that the instantaneous power will exceed x' is:

$$\mathrm{Prob}(xi > x') = x'\ \Sigma\ P_3\ (xi) \qquad (4)$$

Figure 2.6 is a plot of equation 4 in dBmO versus the probability of exceeding that power for a single talker. No compression is applied in this case.

2.5.8.4 m:1 Compression

When compression is applied, the talker power distribution is changed. The mean of -23 dBmO becomes:

((m-1)U-23)/m where U is the unaffected level [7],

and the standard deviation is now 4.8/m. Therefore, in the instantaneous power expression, we should make these changes for a compressed voice. Plots of the cumulative function of the instantaneous power are shown in Figures 2.7, 2.8 and 2.9 for unaffected levels of -33 dBmO, -23 dBmO, and -10 dBmO, respectively.

2750Z/2751Z/2869Z

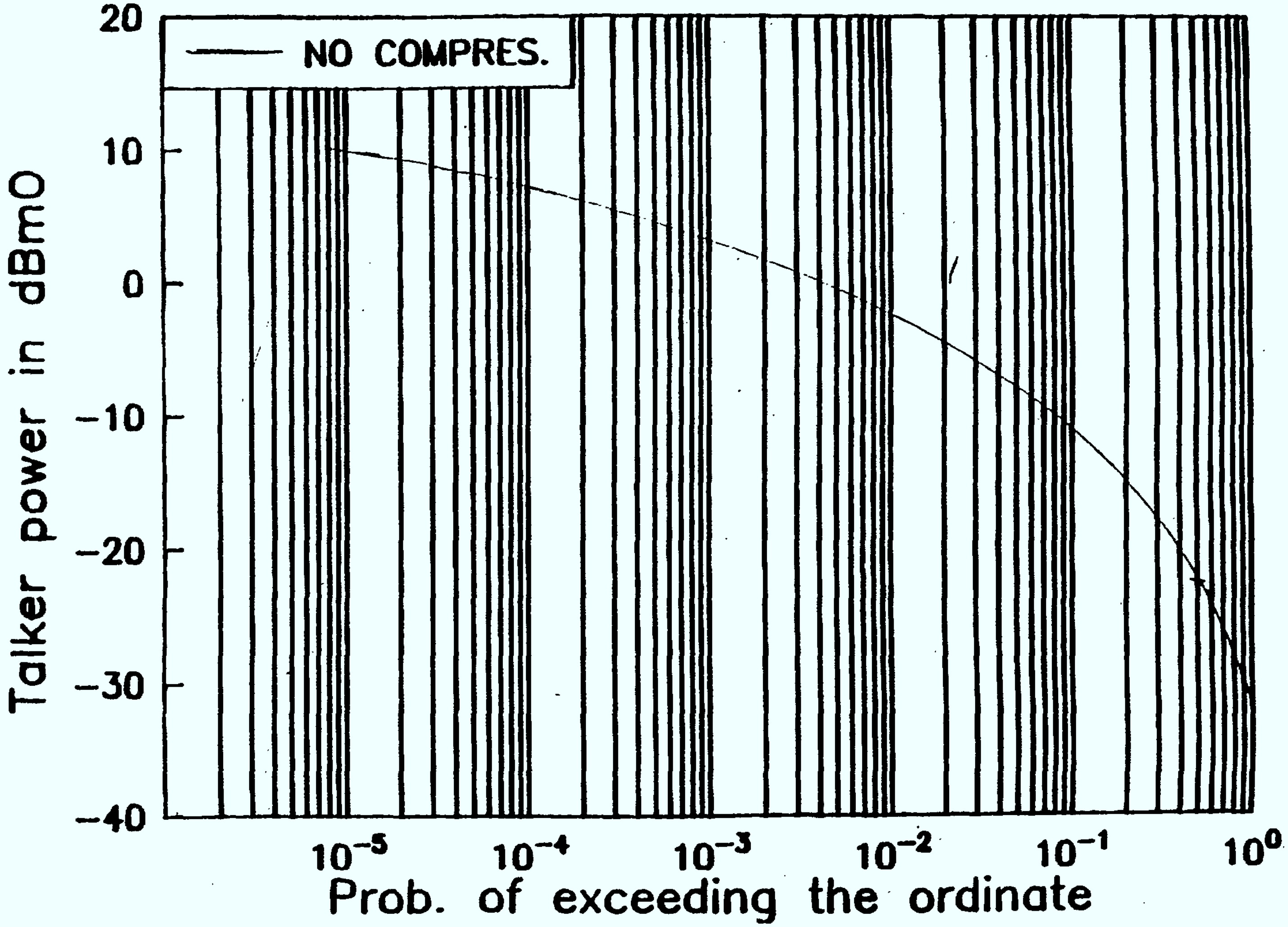

Figure 2.6- The probability of exceeding a power level

With reference to these figures, it is seen that increasing the unaffected level U will also increase the probability of exceeding the power peaks. Note that the 3:1 compression system has less variations, for different unaffected level U, when compared to the 2:1 compression system. This is due to the smaller dynamic range encountered with a 3:1 compression ratio, and to the non-linear aspects of a companded system. In order to limit the power below a certain peak value, it seems that 3:1 companding system is more favorable over the 2:1 system.

An advantage of using a higher compression ratio is the reduction in the dynamic range of the signal. Therefore, less variations will occur on the speech level after the compressor. This is obvious since the standard deviation of the speech power is reduced by a factor of 1/m where m is the compression ratio (see Figure 2.10).

A higher compression ratio means a higher gain for low signal levels. Therefore, the noise level superimposed on the speech will be lower when a high speech level is present. The minimum SNR requirements will limit the compression ratio used. Systems with a high companding ratio are not easy to implement, since it will be more difficult to make the expandor characteristics closely follow the compressor characteristics. The degree of nonlinearity of the compandor depends on the companding ratio. The effect is that more spectrum spreading will occur with a higher ratio.

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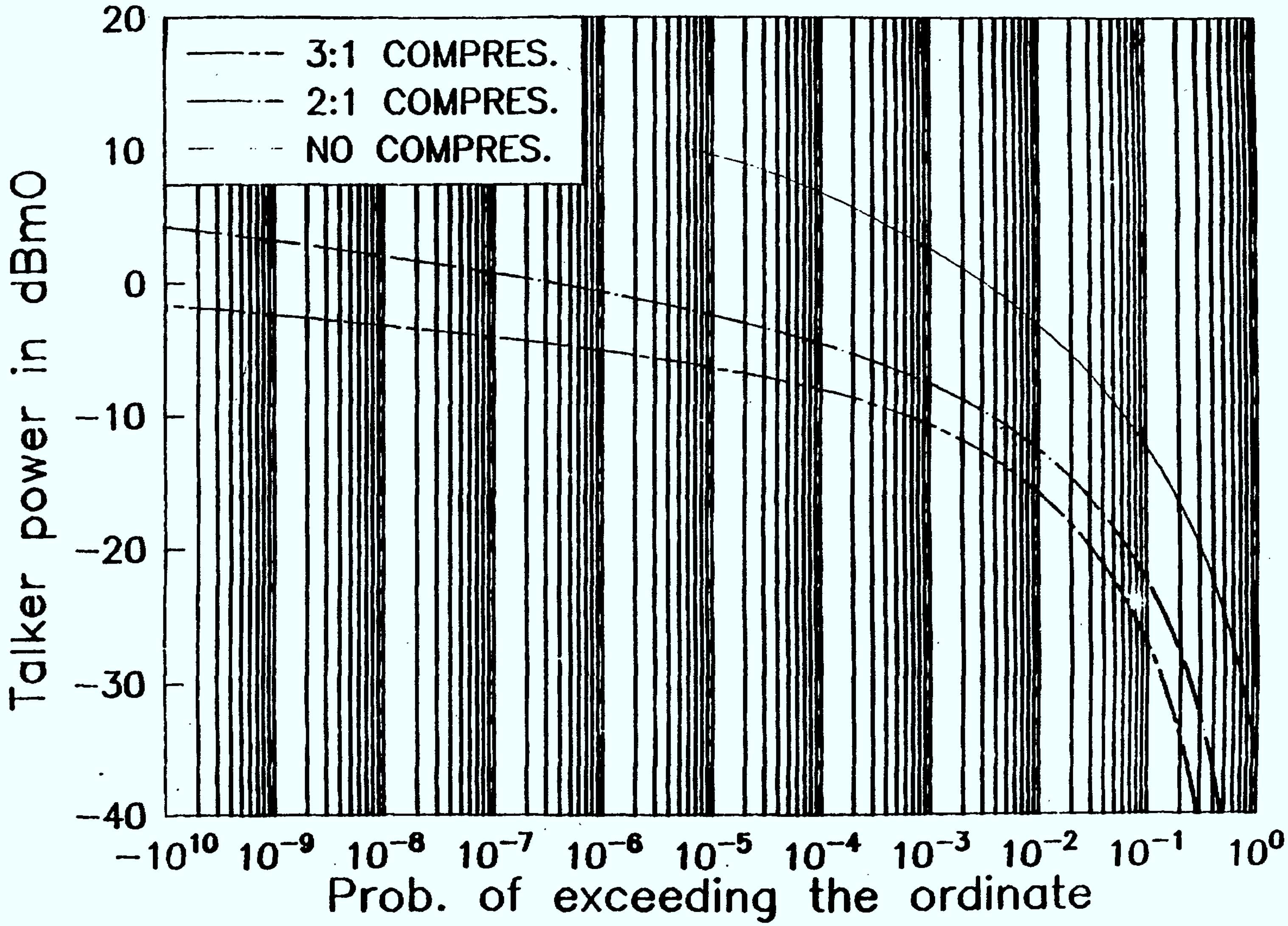

Figure 2.7- The probability of exceeding a power level for companded voice with U= -33 dBm0

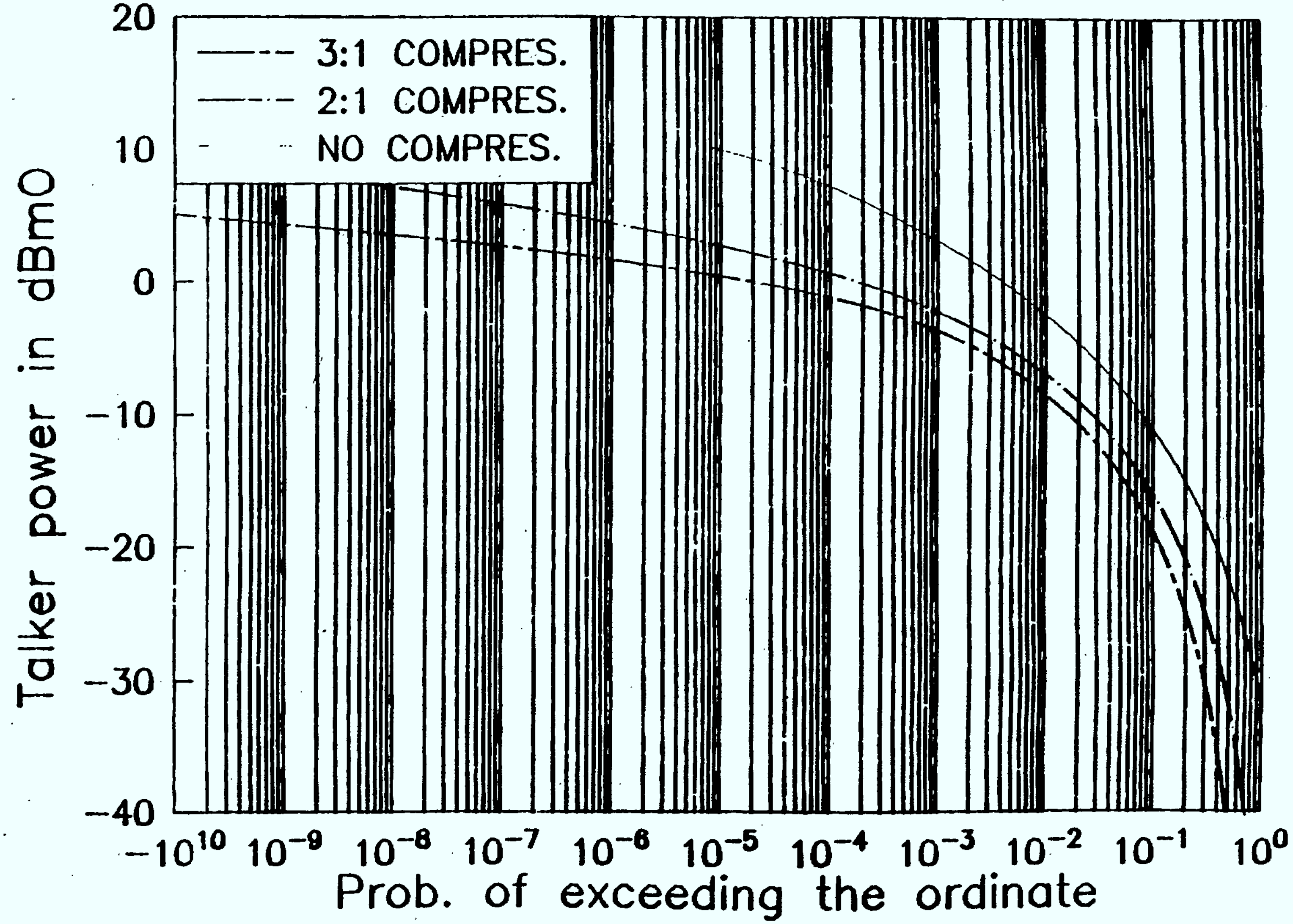

Figure 2.8- The probability of exceeding a power level for companded voice with U= -23 dBm0

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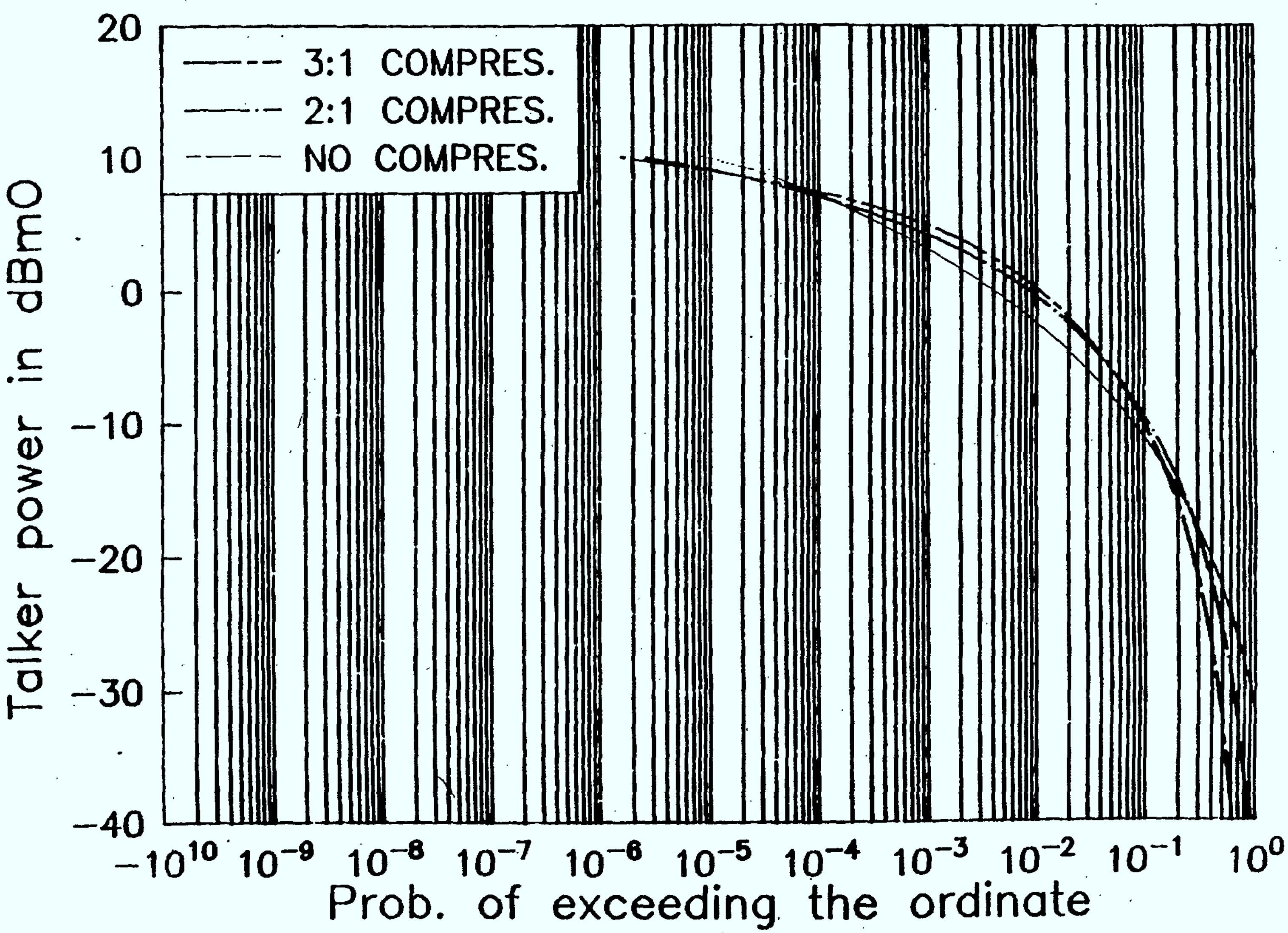

Figure 2.9- The probability of exceeding a power level for companded voice with U= -10 dBm0

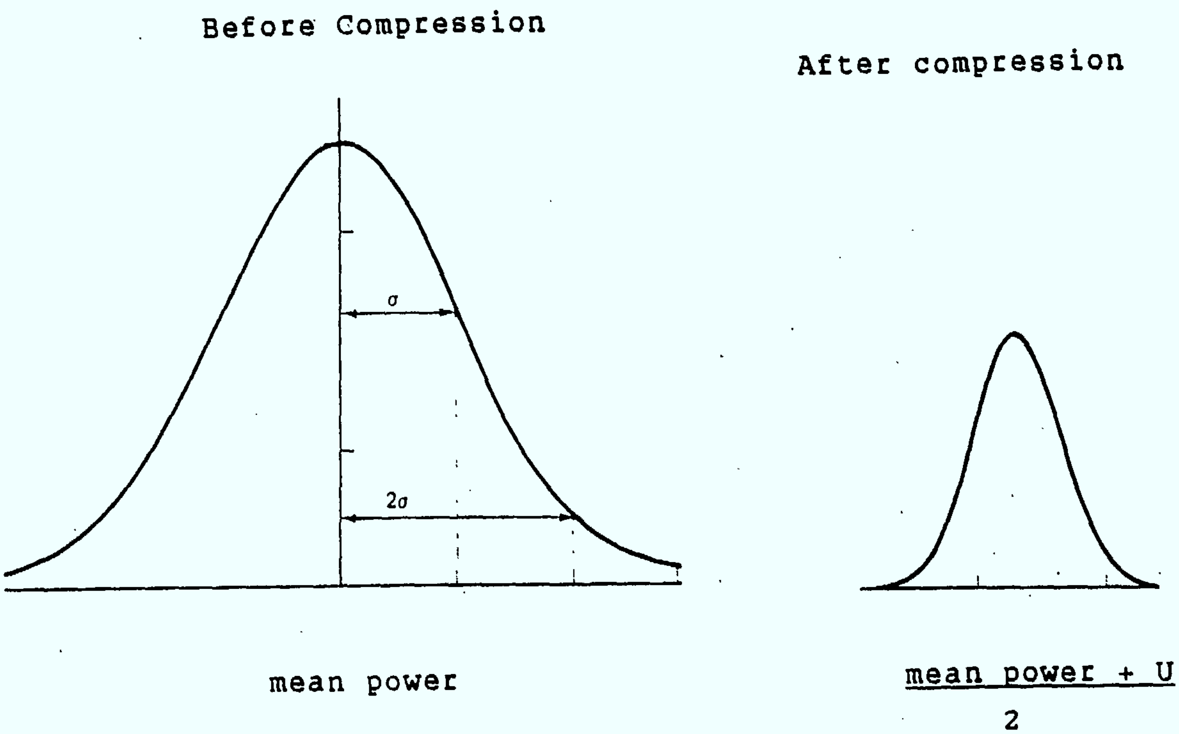

Figure 2.10- Speech power distribution before and after compression

The voice power level of a talker is not a deterministic value. It is always defined with a normal distribution function with given long term average and variance. The behavior of a compressor is different for two types of input signals, such as a tone signal and a voice signal. For the tone signal, which is approximated by a normal distribution with zero variance, if the tone level is lower than the so-called unaffected level, a certain increase in the signal level occurs at the output of the compressor. The compressor operates differently in the presence of speech at its input. It may happen that the output of the compressor is at lower level than its input level even if the average voice power is below the unaffected level. This is explained below.

Let us define the median level of voice as M which is the average talker power, and the variance of speech power as σ^2. Therefore, the long term average level of voice is [9]:

$$A = M + 0.115\ \sigma^2 + 10 \log a \qquad (5)$$

Where a is the speech activity factor. For M = -23 dBmO and σ = 4.8 dB and a = 0.4

$$A = -23 + 0.115\ (4.8)^2 - 4 = -24.35 \text{ dBm0}$$

Consider a 2:1 compressor with an unaffected level U = -20 dBmO. Normally, the -24.35 dBmO level when passed through the compressor should be amplified. The 2:1 compressor transforms the median level M to M' using the transfer characteristics:

$$M' = \frac{U + M}{2} = \frac{-23\ -20}{2} = -21.5 \text{ dBm0} \qquad (6)$$

The standard deviation is halved to 2.4. The new long term average level is:

$$A = - 21.5 + 0.115\ (2.4)^2 + 10 \log a = -24.83\text{dBmO}.$$

Therefore the resulting speech level after the compressor is attenuated.

For the MSAT system, in order to operate at the long term average power, an ALC circuit is introduced before the compressor in order to eliminate the variations in the speech. Thus, the speech, after being processed in the ALC circuit, will have a constant average power and will be similar to a sinusoidal signal at the input of the compressor.

2.5.8.5 Companding Advantage of an m:1 Compandor

On the transmission link, the compressor transforms the input level S_o to a level S_L in accordance with the compressor relationship:

$$S_L - U = (S_o - U)/m, \qquad (7)$$

where U is the unaffected level

Rearranging the previous expression results in:

$$S_L = S_o/m + (m - 1)/m \;\; U \qquad \text{dBmO}$$

Because the active speech is carried 40% of the time, the average speech power C will be - 4dB below S_L. Thus:

$$C = S_L - 4 \qquad \text{dBmO} \qquad (8)$$

The transmission link component of the noise N_T is:

$$N_T = S_L - C/_N -4 \qquad \text{dBmO} \qquad (9)$$

The noise on the transmission link depends on whether the speech is present or not. The resulting "off" noise (when no speech is present) $N_{L\ off}$ on the transmission link is [5]:

$$N_{L\ off} - U = (N_o - U)/m \quad N_{L\ off} = N_o/m \qquad (10)$$
$$+ m - 1/m\ U, \qquad \text{dBmO}$$

Where N_o is the noise present at the input of the compressor. Thus:

$$N_{L\ off} = 10^{(N_o/m + (m - 1/_m)\ U)/10} \qquad (11)$$

The total noise perturbing a speech channel is the sum of N_T and $N_{L\ off}$. That is:

$$N_{off} = U + m(N_{TLoff} - U) \qquad (12)$$

where $N_{TLoff} = 10 \log (10^{NLoff/10} + 10^{\ NT/10})$

$$= 10\ m \log (10^{N_T/10} + 10^{N_{Loff}}) + (1 - m)U \qquad (13)$$

$$= 10\ m \log (10)^{(S_L - C/N - 4)/10} + 10^{(N_o/m + (m - 1/m)\ U)/10)}$$
$$+ (1-m)\ U$$

The compander advantage, CA is the difference between the ratio S_o/N_{off} at the compander output and the ratio S_L/N_T on the transmission link.

$$CA = (S_o - N_{off}) - (S_L - N_T) \qquad (14)$$

$$= S_o - N_{off} - C/N - 4$$

When the speech is present, the noise component due to N_o appearing on the transmission link during the occurrence of active speech is,

$$N_{Lon} = S_L - (S_o - N_o) = S_o + ((m - 1)/m)\, u - S_o + N_o$$

$$= ((1 - m)/m)(S_o - u)) + N_o \qquad (15)$$

when this is combined with the transmission link noise N_{TL}, the total noise is:

$$N_{LRON} = 10 \log (10^{[((1 - m)/m)(S_o - u) + N_o]/10} + 10^{[-C/N - 4]/10)} + S_L \qquad (16)$$

At the output of the expandor, the total noise when the speech is present is:

$$N_{ON} = S_o - (S_L - N_{LRON})$$

$$= S_o + 10 \log [10^{[((1 - m)/m)(S_o - u) + N_o]/10} + 10^{[(-C/N-4)/10]}] \qquad (17)$$

In Figure 2.11, a plot of the companding advantage (equation 14) versus signal power level for different compression ratios is shown.

2.5.8.6 Discussions on Companding

Companded systems have several advantages over non-companded systems. However, a designer should be aware of the problems arising from using a certain compression ratio.

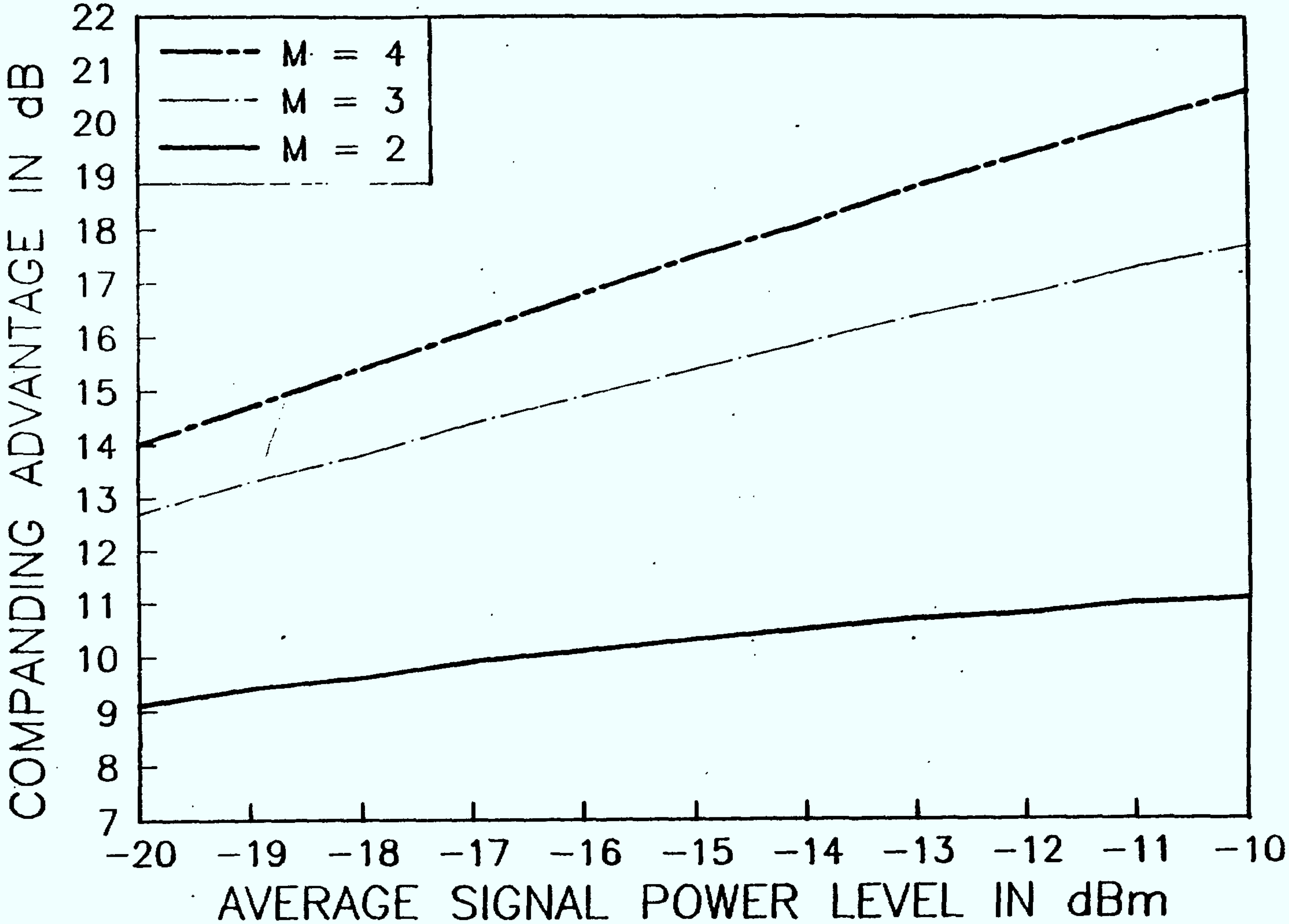

Figure 2.11- Companding Advantage vs. Average Signal Power

The companding ratios of interest are 2:1, 3:1, and 4:1. Since non-linearity problems arise from using compandors in tandem, and since the threshold of a 4:1 compandor is higher than that of a 3:1 compandor, (see Figure 2.12), the 4:1 system is more difficult to design effectively. Both 2:1 and 3:1 systems would operate on a satellite link. The advantages of companding are:

1. Reduction of the dynamic range of the speech signal power and limitation of signal variations over the satellite link.

2. Reduction of the noise and crosstalk on the communication channel.

3. Enhancement of the subjective SNR when speech is absent.

4. Masking the noise when speech is present.

From Figure 2.11 it appears that there is 3 to 6 dB companding advantage, CA, in using a 3:1 system over a 2:1 system depending on the average signal power. However, we know that the compressor and expandor perform two complementary functions and, therefore, the expandor has to closely follow the compressor in order to reduce the signal distortion. Thus, there is a need to define an operating point for the compressor and expandor (see Figure 2.13).

The gain of the compressor and expandor varies with the magnitude of the input signal. Therefore, high gain stability is required. This problem could be solved by choosing the proper attack and recovery time for the compandor. Fast attack time requires a wider bandwidth to transmit the signal, otherwise the reconstructed signal in the receiver will be distorted. On the other hand, a slow attack time will cause an overload, as will be explained in the following. Since the noise sets the compressor gain when speech is absent, if a burst of speech suddenly occurs, then a chance of exceeding the peak level exists. Slow recovery time is unacceptable because the noise between speech pauses will be amplified and become intolerable if the gain does not change sufficiently fast.

The choice of the unaffected level, U, of the compandor has some influence on the power overload. When U is greater than the average speech power the probability of exceeding the peak power is higher. From Figures 2.6, 2.7, 2.8 and 2.9, the 3:1 companding system seems more stable than the others. In general, the factors that determine the compression ratio are:

- The minimum SNR requirement.

- The system complexity. (It is known that a higher companding system is more complex to design).

- The degree of non-linearity. More spectrum spreading occurs at higher compression ratios. This would seem undesirable on the MSAT channels, since the adjacent channel interference should be kept at an acceptable level.

In summary, some words of caution on the average speech power are warranted here. In the past, it had always been assumed that the average speech power was -15 dBmO. However, detailed traffic monitoring studies revealed that there had been a long-term secular change in the average per channel talker level of telephone users in North America. This is believed to result from improvements in system quality and more sensitive telephone end instruments, both of which permit the average talker to obtain satisfactory performance without speaking as loudly as before. The new average talker level per channel is found to be about -23 dBmO. However, in a mobile environment the user would tend to speak louder and thus the average speech power should be higher. Due to the absence of data on this matter, it is recommended that -15 dBmO be used for average speech power until more information becomes available. There exists some trade-offs between companding advantages, channel efficiency, and acceptable level of distortion in order to obtain a certain degree of intelligibility in the power and bandwidth limited MSAT System.

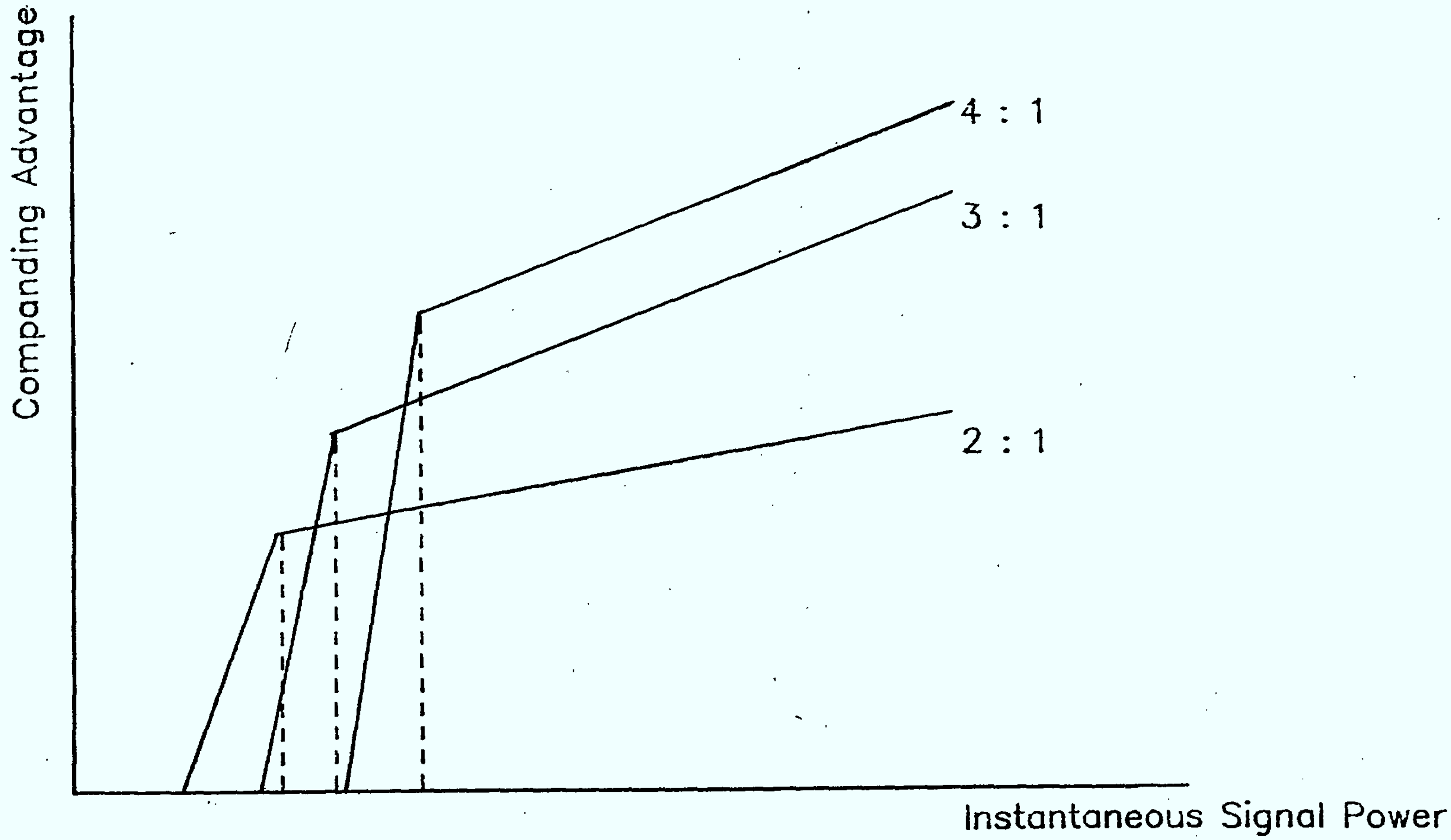

Figure 2.12 : Companding Advantage vs. Signal Power

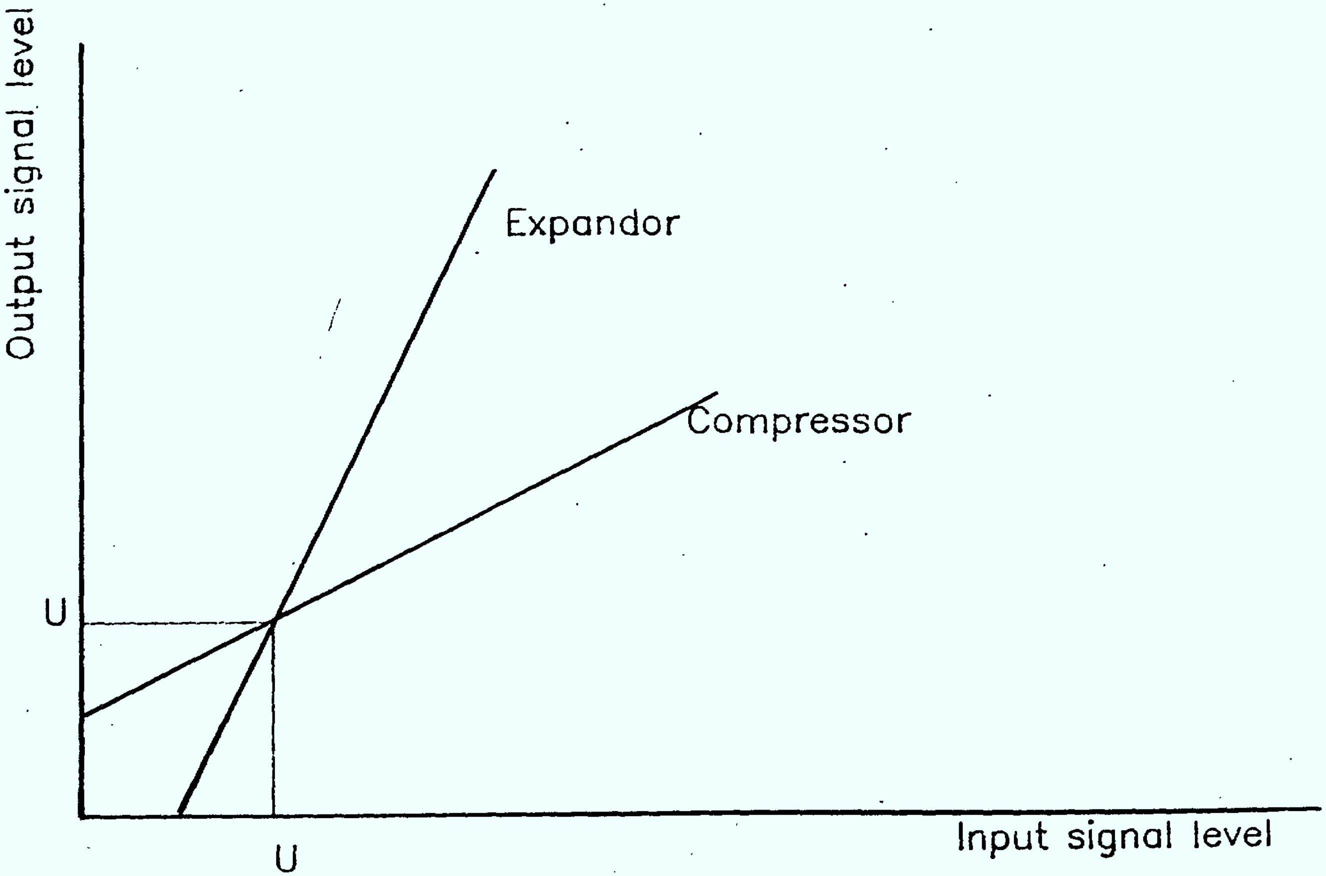

Figure 2.13 : Compressor/Expandor characteristics

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CHAPTER III - DATA TRANSMISSION APPLICATION TO MSAT

3.1 Introduction

The prime objective of the MSAT system is to provide a voice communications service to mobile terminals operating in Canadian remote areas which cannot be served economically by conventional mobile systems. Recently, it has been perceived that a substantial market for mobile data services could also exist. In this chapter, we study the technical requirements for data transmission that can be integrated into the MSAT system. It is expected that data services are implemented in a manner that permits the use of low cost terminals. However, due to the multipath fading and shadowing environment, the high quality performance required for data transmission is not easy to obtain. A portion of the data communications service will be between fixed terminals which require a smaller fade margin and thus the terminal cost might be less than that of a mobile terminal. Also, for a terminal with both voice and data communications capability, the sharing of terminal equipment between voice and data tends to reduce the cost.

3.2 Multiple Access Techniques

The multiple access technique has particular implications on the terminal configuration, particularly the High Power Amplifier (HPA) and channel equipment required. In choosing the data service protocol, the following parameters are to be considered:

- packet length,
- half duplex or full duplex transmission,
- required message transmission performance, i.e. the probability of error and delay, and
- C/N_o variations (fading, interference, etc.)

The number of mobiles transmitting on the reverse link is expected to be much greater than the number of base stations transmitting on the forward link, whereas the traffic volume in the forward link could be higher than that of the reverse link. Therefore, the access method for the reverse and forward links may be different.

Response time requirements are important in the selection of a multiple access technique. A fixed channel assignment achieves a fast response time, but is inefficient in utilizing satellite bandwidth. The fast response required by MSAT rules out polling and carrier-sensed techniques.

3.3 Transmission Rate

The user is ultimately concerned with the transfer rate of information bits. The actual transmission rate is usually higher than the transfer rate due to the factors such as overhead bits, coding and burst operation.

For the data services application, a lower limit is placed on the transmission rates by the effect of frequency uncertainties (e.g. phase noise) on the minimum C/N_o. This lower transmission rate depends on the modulation technique employed.

In choosing the transmission rate, an important factor is the capacity requirements of a particular data service. Considering PSK, MSK and FSK, the 5 kHz MSAT channel is suitable to practical transmission rates up to 4800 bps [5]. The optimum transmission rate, in terms of bandwidth efficiency and complexity is 2400 bps. However, a data service may not have sufficient traffic or power to support a transmission rate of 2400 bps. In this case, lowering the transmission rate proportionally may match the users requirement.

3.4 Modulation Techniques

In the selection of an appropriate modulation technique, several factors must be taken into account. These include:

- BER performance
- bandwidth efficiency
- acquisition time
- the effect of frequency uncertainties and phase noise
- the effect of multipath fading
- implementation complexity

One of the key decisions is the choice of detection type at the desired transmission rate. Based on the above factors, binary coherent PSK, MSK and differentially detected MSK appear to be the candidates in most cases.

Coherently detected systems have excellent performance in the presence of gaussian noise, however they are not tolerant of Doppler shift, shadowing and multipath fading [6]. These effects are very important in mobile communications, because mobiles frequently lose synchronization due to multipath and shadowing. In these cases rapid re-synchronization is required.

Therefore, coherent detection is mainly applicable for the transmission of long messages on the return link only if maintaining carrier phase synchronization is not difficult. Otherwise, differential detection is more suitable than coherent detection.

FSK is also considered for some applications; this technique has advantage when the transmission rate is low and the phase noise is significant.

Automatic frequency control (AFC) is necessary to compensate for carrier offsets introduced by the various conversion oscillators and Doppler shifts appearing in the system. AFC requirements depend on the selection of both the digital modulation technique and the transmission rate. Frequency tracking capabilities are limited only by C/N.

Telesat is currently carrying out further investigation on the types of the digital modulation technique suitable for MSAT services. A report will be available in due time.

3.5 Channel Fading

Assuming a 2400 bps transmission rate, the channel can be considered as slowly fading [5] if:

- packet sizes are less than 500 bits,
- vehicle speed is less than 50 km/hr, and
- the mobile is operating in an environment characterized by infrequent shadowing

Note that the probability of error can be improved by repeating the packet.

On the other hand, the channel is considered as fast fading if [5]:

- packet sizes are over 200 bits and
- vehicle speed exceeds 90 km/hr and
- the mobile is operating in an environment characterized by heavy or medium shadowing (e.g. shadowing caused by trees).

A forward error correction scheme can be used to improve the performance of a fast fading channel.

3.6 Error Control

The necessity of detecting errors in transmitted packets requires the use of an error control scheme capable of detecting and correcting errors of a bursty nature. In evaluating the suitability of a given error control scheme, the following factors must be taken into account:

- Modulation technique
- Channel statistics
- Coding gain required
- Bandwidth required to accommodate the coding bits
- Message length
- Decoder complexity (cost).

The most common method of predicting the presence of errors in a message is by error detection coding.

Automatic Repeat Request (ARQ) systems are error control strategies whereupon one or more retransmissions may be required in order to correctly send a message. The receiver must be able to determine when errors are present in a received message, and have access to a return communication link to inform the sender of the status of the message received. Three types of acknowledgement (ACK) schemes exist;

i) Positive Acknowledgement Scheme:

Whenever the receiver receives a packet without errors, it sends an acknowlegement back to the transmitter. If a packet is received with errors, no message is sent back to the transmitter. If the transmitter does not receive an acknowledgement from the receiver, it infers that the original message was lost or received with errors and it will retransmit the message after waiting a suitable length of time. The advantage of this method is that there is a small probability that a packet is lost.

ii) Negative Acknowledgement Scheme

The receiver sends a negative acknowledgement whenever a packet is received with errors. If a message is received correctly, no acknowledgement is sent. The advantage of this method is that less messages are sent from receiver back to the transmitter, but there is a risk that a packet can be lost without the transmitter realizing it, since if no message is received back from the receiver, it is interpreted as successful reception.

iii) Positive and Negative Acknowledgement

In this case, the receiver sends back a message to the transmitter for every packet it receives. If a packet is received without errors, a positive acknowledgement is sent. If a packet is received with errors, a negative acknowledgement is sent. If the transmitter does not receive any kind of a message back from the receiver, it assumes that the packet was lost and retransmits it. This method minimizes the delay in retransmitting packets incorrectly received as compared with the positive acknowledgement scheme. It has a disadvantage that very many messages are transmitted from the receiver to the transmitter.

3.7 Digital Modulation Techniques Comparison

The studies carried out by now have suggested the use of differential minimum shift keying (DMSK) for digital voice and data modulation as well as for the DAMA interface. This choice was based on the properties of DMSK, which yield some advantages over many other candidates such as QPSK, DPSK, OK-QPSK, and GSMK. Any modulation, based on amplitude variations, will have a degraded performance in the presence of fading in a mobile communications environment. Therefore, a modulation scheme with constant amplitude or a constant envelope should be used. In the following sections, we will compare the modulation techniques cited above, in terms of error-rate performance, spectral efficiency, performance in the presence of thermal noise, adjacent channel interference, co-channel interference, phase noise, bandwidth limitations and phase distortion. Note that the choice of modulation is also dependent on the Doppler effects, acquisition time, system frequency error, complexity, and cost.

3.7.1 Error-Rate Performance

QPSK provides the same error-rate performance as antipodal signalling but halves the bandwidth occupancy as compared with BPSK. The modulator output signal of a QPSK is the summation of two BPSK signals, both operating at the same transmission rate but with orthogonal carriers. Those two components are referred to as the I and Q channels. Independent signalling occurs in each channel, and each channel will have an error rate equal to that of BPSK. The total output power per channel is half that of a BPSK signal, and the effective receiver noise bandwidth is halved because of the division of the input bit stream into I and Q components. Therefore, the same energy per bit to noise density ratio (E_b/N_o) is maintained. The probability of error is [2]:

$$P_b = 1/2 \text{ ERFC } [E_b/N_o]^{1/2}$$

where ERFC [.] is the complementary error function.

Offset QPSK is a form of QPSK in which the digits in the quadrature channels have a relative delay in their transitions. If the serial input data has a duration T, then the I and Q data will each have duration 2T. The relative delay between both channels is T. The purpose of this delay is to restrict the carrier phase transitions from having 180° phase transitions. When filtered, the offset QPSK signal will have less envelope fluctuations compared with QPSK. In the unfiltered case, the introduction of a delay has no performance effect and offset QPSK has the same error rate performance as does conventional QPSK.

Fast frequency shift keying, referred to as minimum shift keying (only in the case of differential detection) is a digital frequency modulation scheme with a modulation index of 0.5. The term fast is used because more bits per second can be transmitted in a given channel bandwidth compared with BPSK. For a modulation index of 0.5, the best error rate performance obtained will be the same as for BPSK. MSK can also be viewed as a QPSK type signal where the pulse shape of each symbol is a half-period sinusoid.

The differential phase shift keying signal is an encoded BPSK waveform for which the receiver avoids the need for provision of a phase-locked carrier. The DPSK error rate is [2]:

$$P_{DPSK} = 1/2\ e^{-E_b/N_o}$$

GMSK is a form of MSK where the baseband filter has a Gaussian shape. This technique yields a constant envelope and narrowband modulation suitable for power and spectrum limited digital communications. The premodulation LPF meets the following conditions [8]:

- Narrow bandwidth and sharp cut-off to suppress the higher frequency components.

- Lower overshoot impulse response to protect against excessive instantaneous frequency deviation.

- Preservation of filter output pulse area to assure the phase shift of $\pi/2$ for coherent detection capability.

The optimum 3 dB bandwidth of the gaussian filter is proved theoretically to be BT = 0.5887 where B is the 3 dB bandwidth and T is the bit duration. This degrades the required E_b/N_o by 0.15 dB [9]. This will yield a higher probability of error than that of MSK without Gaussian filtering.

Another form of modulation, developed by Skylink, is the rectangular spectrum modulation (RSM) [7]. In this technique, IF equalization is done on the power spectral density of PSK signals. It is known that the power spectrum of PSK signals has the sin X/X shaping. In order to render this spectrum rectangular (constant envelope), a multiplication by X/sin X is done. The spectral efficiency of RSM is higher than that of PSK [7]. Figure 3.1 shows the probability of error versus E_b/N_o for different modulation schemes.

3.7.2 Spectral Efficiency

The spectral efficiency represents the bit rate that could be contained in a 1 Hz bandwidth [1]. The spectral efficiencies of different modulation techniques are shown below [1]:

Modulation	Spectral Efficiency, b/s/Hz
QPSK	2
OK-QPSK	2
DPSK	1
MSK	1.7 approximately
BRSM	1.2
QRSM	2.4

3.7.3 Power Spectral Density

The power spectral density of PSK systems is given by [2]:

$G(f) = P/Rx \quad (\sin^2(f\pi/Rx))/(f\pi/Rx)^2)$

where P is the modulated signal power and Rx the symbol rate.

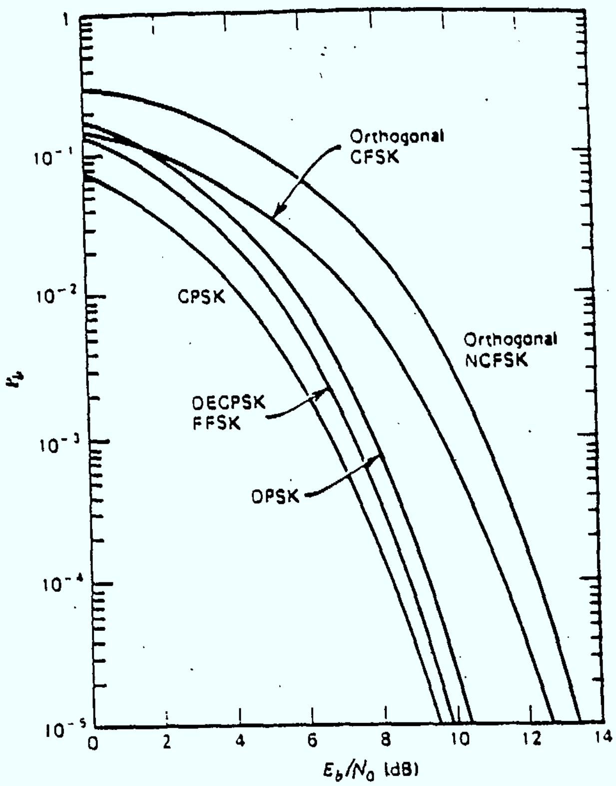

Figure 3.1: Probability of a Bit Error for Various Modulation Techniques

In the case of a BPSK signal, Rx = bit rate = R, while for QPSK, Rx = R/2, which means that QPSK occupies one-half of the bandwidth required for BPSK. The spectral shape is the same for both, but in the case of QPSK, it is compressed by a factor of 1/2. With QPSK, the main lobe contains 92.5% of the power and spectral nulls are at $\pm 1/2$ times the bit rate from the carrier centre frequency. The first sidelobe roll-off is 6 dB/octave. Offset QPSK has the same spectrum as QPSK.

In the case of MSK, the power spectral density is given by [1], [2]:

$$G(f) = 8P_c/\pi^2 R\ (1 + \cos 4\pi f/R)/([1 - (16/R^2)f^2]^2)$$

where R is the bit rate and P_c is the carrier power.

The main lobe contains 99.5% of the power and spectral nulls are at ± 0.75 times the bit rate from the carrier. The first sidelobe is 25 dB below the main lobe and the spectral roll-off is 12 dB/octave.

The out-of-band power G_{ob} is given by:

$$G_{ob} = 1 - (\int_{-B}^{B} G(f)\ df)/(\int_{-\infty}^{\infty} G(f)\ df)$$

The equal power bandwidth occurs at an offset from the carrier frequency of 0.76/T. Closer to the carrier, QPSK has more relative power while beyond the cross-over point MSK contains more power. Figure 3.2 shows the power spectral density of both QPSK and FFSK while Figure 3.3 is a plot of the out-of-band power versus the filter 3 dB bandwidth for both QPSK and FFSK.

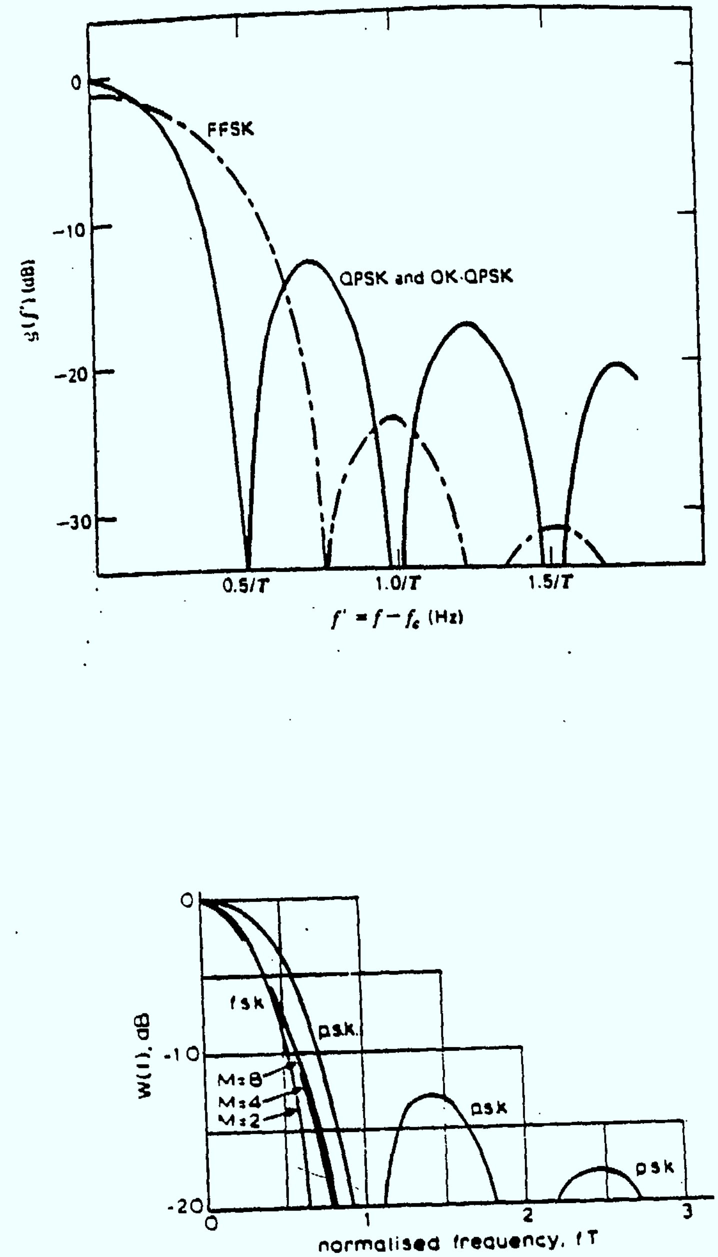

Figure 3.2: Spectra of Various Modulation Techniques

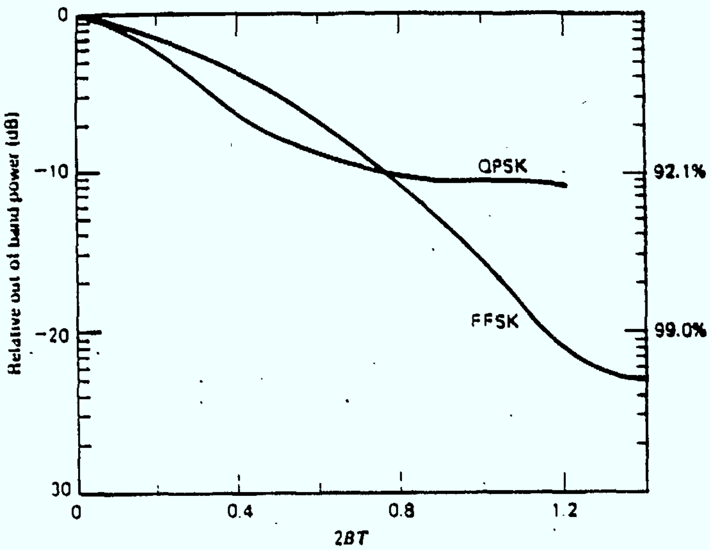

Figure 3.3: Relative out-of-band power for QPSK and FFSK versus filter bandwidth

3.7.4 Adjacent Channel Interference

Adjacent channel interference refers to the degradation in performance in one channel arising from spillover from an adjacent channel. The main lobe of MSK is wider than for QPSK, but the sideband roll-off is sharper. The following table shows the proximity with which channels can be spaced for no more than 1 dB E_b/N_o degradation [2]. The spacing is given as a function of the relative power of the interfering channel to the main channel.

Modulation	Relative Interference, 0 dB	+10.0 dB
QPSK	4.5 R	13.5 R
OK-QPSK	5.0 R	14.0 R
MSK	1.5 R	2.5 R

Table 3.1: Centre Frequency Separation for Eb/No loss of 1 dB maximum.

It is seen that MSK is superior to QPSK in terms of adjacent channel interference. This means that the channel can be spaced closer and less control of transmit power is required. In a multiuser environment with significant uplink power variations, MSK appears to be attractive.

3.7.5 Co-Channel Interference

Co-channel interference refers to the degradation caused by an interfering waveform appearing within the signal bandwidth. The interferer could be either an unmodulated carrier or a low-power carrier modulated, for example, by the same type of modulation for the case of MSAT. For MSK, an interfering carrier 15 dB below the wanted carrier would cause a degradation of no more than 1 dB in the E_b/N_o at a bit error rate of 10^{-5}. For the same probability of error, 2 dB degradation occurs in the case of QPSK [2].

3.7.6 Phase Noise

Regeneration of the carrier at the receiver implies that the local carrier reference is imperfect and is a noisy phase reference. In the case of phase difference between the carrier and the received signal, the I and Q channels are reduced in level and are not mutually independent.

In the case of QPSK, the alignment of bits is the same in the two channels and a constant interference results over the I-channel bit interval. With offset QPSK, during a bit duration in the I channel, a data transition in the Q channel may occur so that the average interference is reduced. MSK is similar to QPSK, but the interference is reduced due to the continuous phase characteristics.

3.7.7 Bandwidth Limitations

The modulated signals are filtered before the process of transmission takes place. The bandpass filter used has certain amplitude and phase characteristics which lead to degradations in performance. The net effect is pulse smearing and amplitude fluctuations result. The envelope fluctuations induced in the modulated signal become important when the signal is passed through a non-linear amplifier which has an AM/PM transfer characteristic that causes additional distortion. If the filter is not symmetrical about the centre frequency, mutual interference occurs between the I and Q channels.

Since the main lobe for MSK is wider than that of QPSK, a filter with bandwidth of 0.9 times the data rate would cause about 1 dB degradation in the E_b/N_o in the MSK case [3]. The degradation is more severe when using higher order filters.

3.8 Effect of Band and Amplitude Limiting on the E_b/N_o Performance of MSK

Filtering will cause a degradation in the E_b/N_o. In the absence of bandwidth and amplitude limiting between the modulator and the demodulator, the E_b/N_o of MSK is identical to that of coherent BPSK [4]. Band-limiting causes Intersymbol Interference (ISI) on the data carried by each of the quadrature phases. Amplitude limiting will cause an interphaser crosstalk. For systems where the ratio of link bandwidth to link rate is greater than 0.6, the E_b/N_o degradation is less than 1 dB [4].

The amplitude limiter increases the level of the sidelobes beyond the main lobe. It is found also that 99% of the power is contained within a bandwidth of 1.1 times the link rate. E_b/N_o degradation, due to amplitude limiting ranges from 0.3 to 0.5 dB [4].

3.9 Co-Channel Interference Effect on E_b/N_o for a MSK Modem

In a digital communication system, the carrier power is related to the energy per bit as follows:

$E_b/N_o = C/N_o \bullet 1/f_b$ where f_b = bit rate

For MSAT, f_b = 2400 bps and BW = 5000 Hz. The power spectrum of an interfering signal will superimpose on that of the wanted signal. The interfering carrier power is defined with respect to the wanted signal power. In the following we will compute the amount of degradation in the E_b/N_o performance when different co-channel interferers are disrupting the channel.

From the Bit Error Rate (BER) characteristics of MSK we have:

BER (Pe)	E_b/N_o [dB]
10^{-2}	7.3
10^{-3}	9.7
10^{-4}	11.4
10^{-5}	12.7
10^{-6}	13.7

Table 3.2: BER versus Eb/No for MSK.

C/I is defined as the carrier-to-interfering carrier ratio. For example, a C/I ratio of 15 dB means that the interferer's power is 0.0316 of the carrier power. At a BER of 10^{-3}, E_b/N_o = 9.7 dB, and then, for the worst case situation of 5000 Hz channel bandwidth we have,

$$C/N = 9.3325 \bullet 2400 \text{ bps}/5000 \text{ Hz} = 4.4796$$

By making a simplifying assumption of a flat spectrum for the interfering signal, the total noise power is:

$$N + I = (1/4.4796)\ C + 0.0316C = 0.2548C$$

Therefore, C/(N + I) = 5.94 dB; and the corresponding E_b/N_o is then 9.125 dB.

The same procedure as above has been carried out for different BER and the E_b/N_o degradations are shown in the following table:

BER	E_b/N_o [dB]	E_b/N_o Degradation [dB]
10^{-2}	7.0	0.3
10^{-3}	9.1	0.6
10^{-4}	10.6	0.8
10^{-5}	11.6	1.1
10^{-6}	12.4	1.3

Table 3.4: BER versus E_b/N_o for MSK in the presence of a cochannel interferer.

Figure 3.4 shows the degradation in the BER in the presence of a co-channel interference with two different C/I ratios.

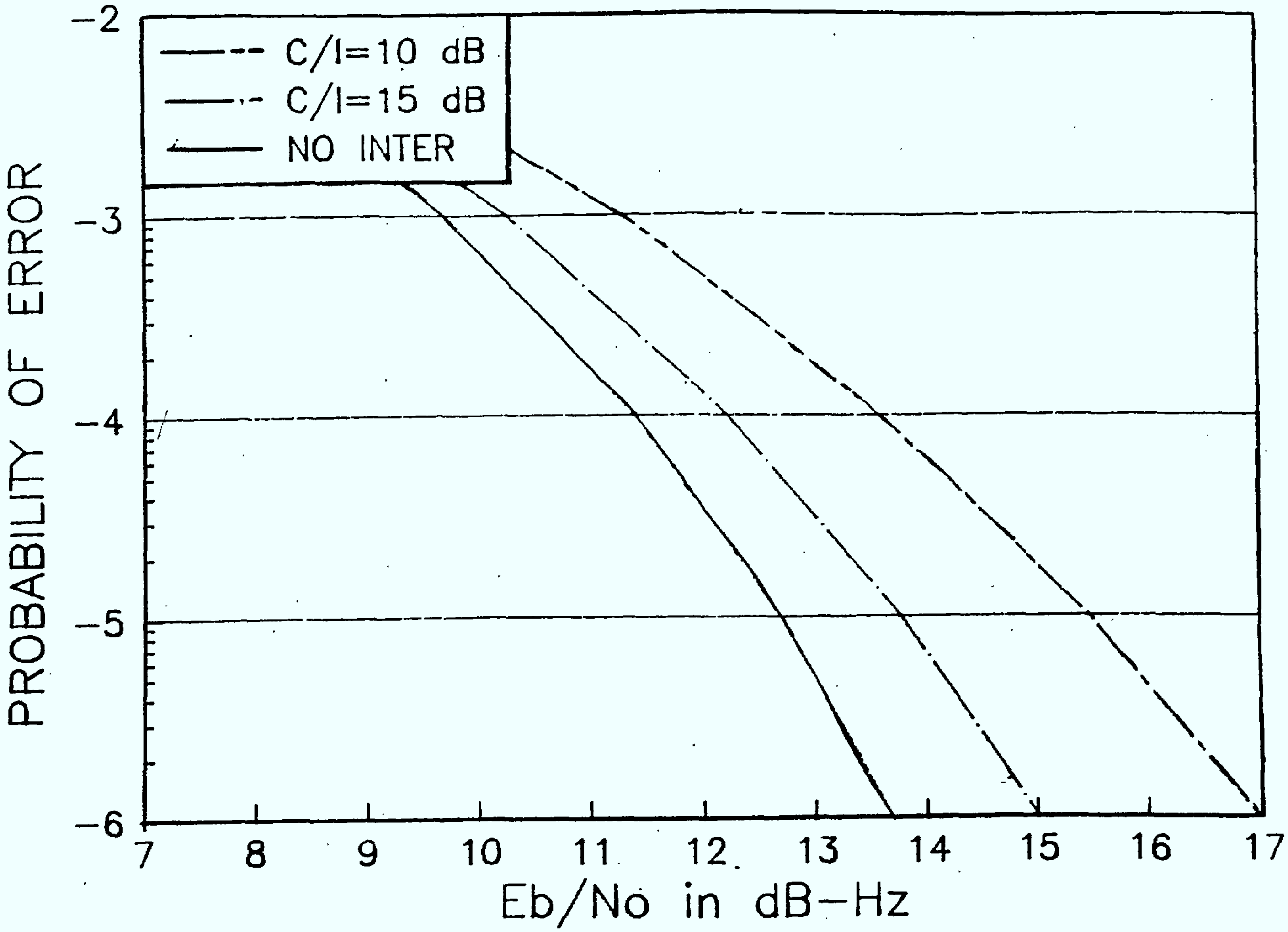

Figure 3.4- BER performance of DMSK in the presence of a cochannel interferer

3.10 The Error Floor

The error floor associated with the 2400 bps signaling rate in a high frequency band such as 800 MHz poses an obstacle for mobile communications via MSAT channels when the line of sight signal is attenuated. Often, frequency efficiency must be greatly sacrificed to restrict this floor below a desired threshold. The existence of a strong line of sight signal is helpful in controlling the link's error rate. The error floor of the DMSK receiver for different line-of-sight-to-multipath ratios (LMR) for a Rician channel, is shown in Figure 3.5, with a Doppler frequency shift of 72 Hz [6]. This figure shows the irreducible error rate as a function of LMR. It should be recognized that a large LMR, i.e. larger than 13 dB, constitutes a reasonably stable link, whereas a small LMR constitutes a heavily faded link. Note that the error floor required for voice communications is 10^{-3} [6].

3.11 Frequency Tracking

A combination of the Doppler shift and the local oscillator uncertainty gives rise to frequency errors with a magnitude too large to be remedied by simple techniques. Therefore, a tracking loop that can correct large frequency errors is required for stable and efficient operation of the receiver. The structure of the receiver at baseband lends itself to incorporation of an AFC loop without substantial added complexity. The block diagram of the receiver with the AFC loop is shown in Figure 3.6 [6]. Figure 3.7 shows the BER performance of DMSK with and without the AFC

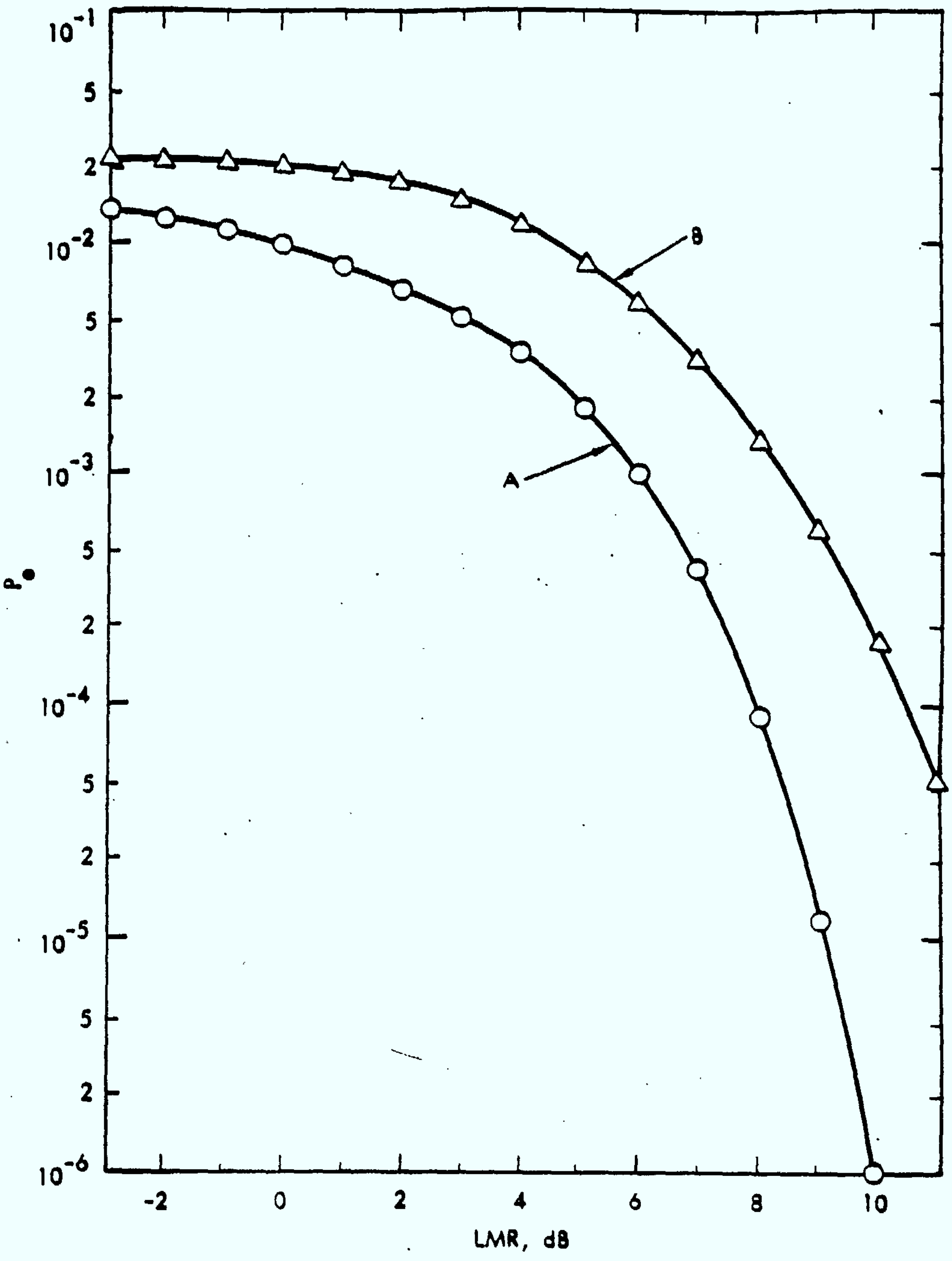

Figure 3.5- The error floor as a function of LMR in a mobile satellite link with $f_d = 72$ Hz. Curve A was obtained using the conventional model, and curve B was obtained using the worst case model.

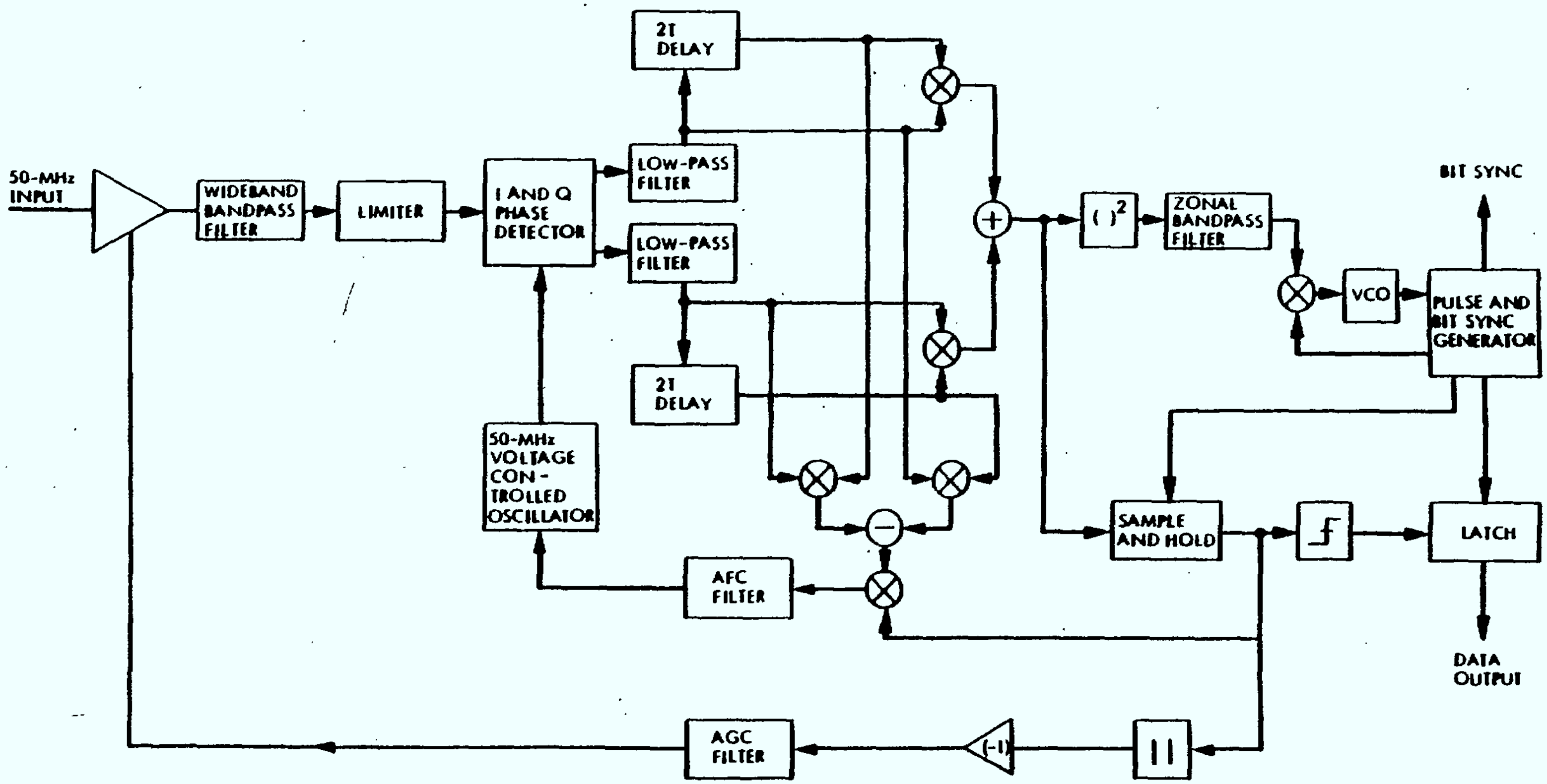

Figure 3.6- The two-bit DMSK receiver with the AFC,AGC and bit synchronization loops in operation

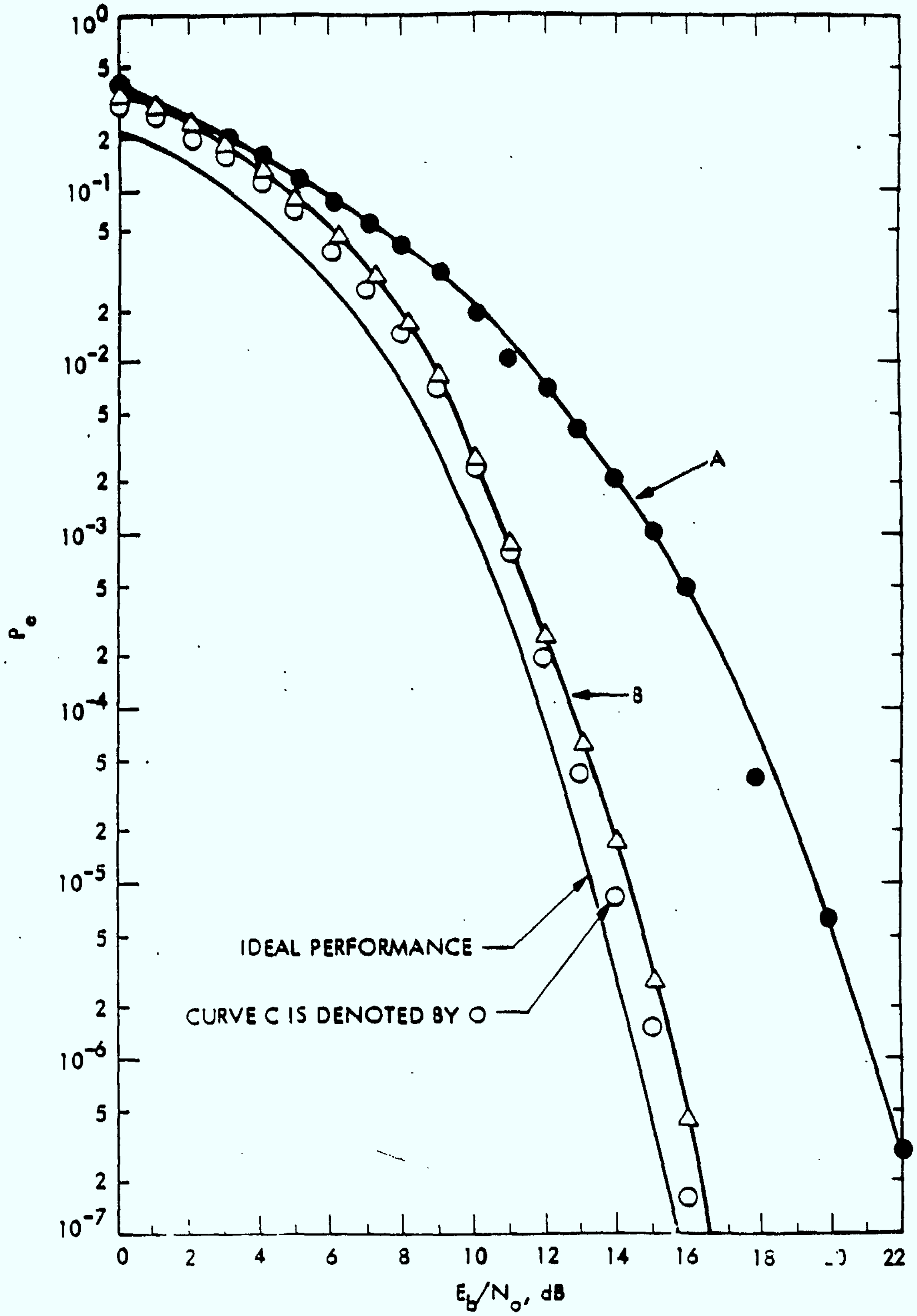

Figure 3.7- The receiver error performance in the presence of a frequency error of 100 Hz (curve A),with AFC(curve B), and with AFC in the absence of a frequency error (curve C)

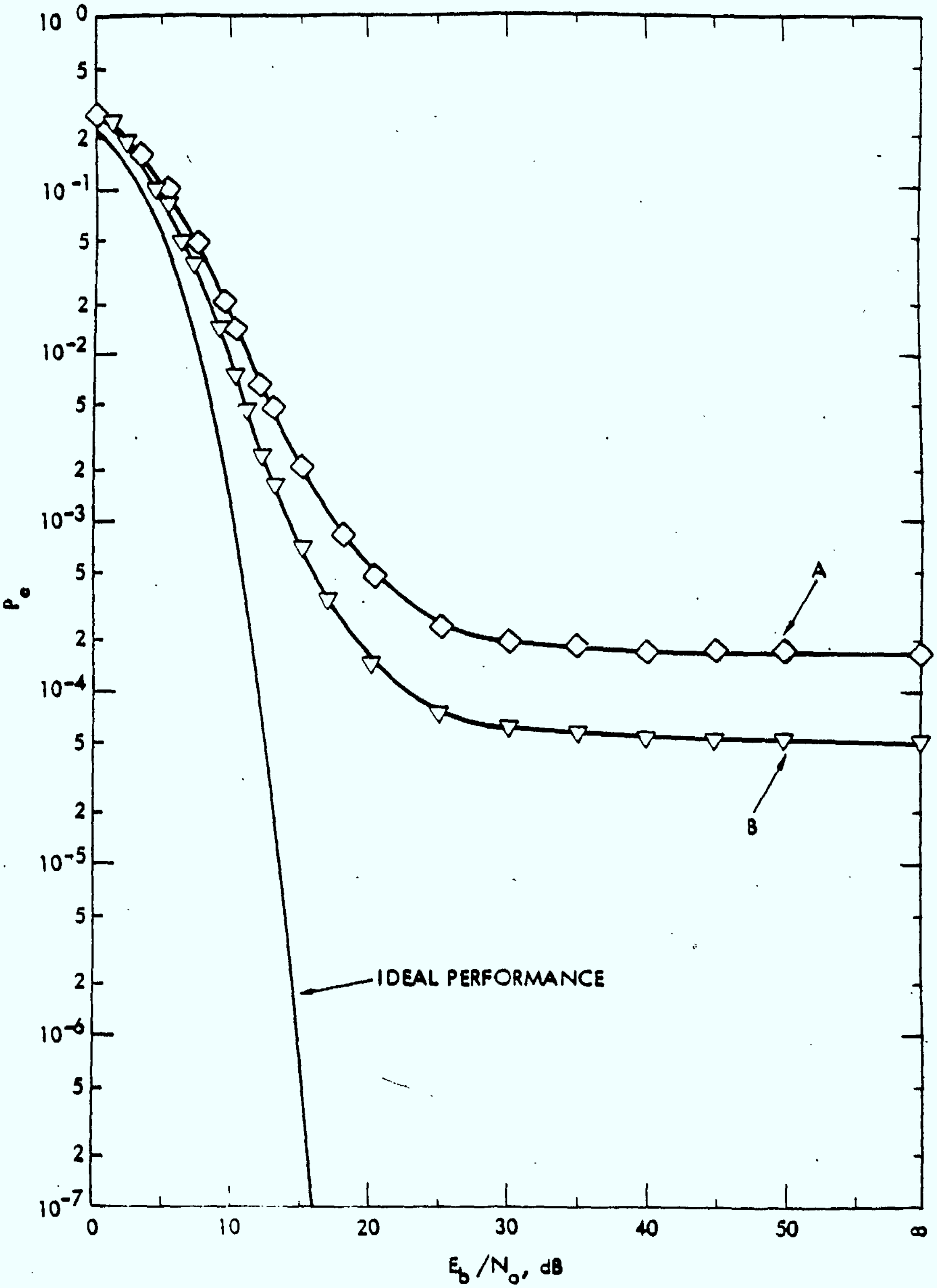

Figure 3.8- The receiver performance under the worst case Rician model with LMR= 10 and f_d =72 Hz. Curve A was produced without the AFC loop, and curve B with the AFC loop.

loop and in the presence of a frequency error of 100 Hz, while Figure 3.8 represents the worst case Rician model with a Doppler shift of 72 Hz. From Figure 3.7, we see that there is 5 dB improvement at 10^{-4} resulting from the use of an AFC loop.

3.12 Summary

At the present time it appears that DMSK is the most favourable candidate to operate in a mobile satellite system at a bit rate of 2400 bps under MSAT fading and shadowing environments. That is mainly due to:

- Modem implementation simplicity,
- good error rate performance comparable to coherent BPSK,
- its spectral efficiency,
- minimal degradation due to phase noise

The minimum performance requirements for DMSK modem will be presented in Chapter IV.

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CHAPTER IV - SPECIFICATIONS FOR MSAT UHF TERMINALS

4.1 Introduction

The MSAT mobile terminals will provide mobile radio and telephone services to suburban and remote areas. Call placement and reception will be automatic; the user's operating procedure will be similar to the cellular systems.

This chapter provides specifications for ACSSB half duplex UHF MSAT mobile terminals. The specifications given here define the overall, end-to-end performance of the unit. The subsystem parameters are not given here since it was felt that manufacturers should have the freedom of selecting the system parameters of their preference as long as the end-to-end requirements are met.

Note that the specifications given in this chapter are based on Telesat's current system design, which has not yet been finalized, as well as our present understanding of the related technologies. Therefore, changes in the Telesat system design or related technological advances may result in revision of these specifications in the future.

Appendix A provides the rationale behind some of the specifications given in this chapter.

4.2 Specifications for MSAT UHF ground terminals (ACSSB)

Amplitude Companded Single Sideband (ACSSB) is used for analog modulation of a voice signal. The terminal transmit frequencies at UHF will be 821 to 825 MHz while the receive frequencies will be 866 to 870 MHz. The transmit and receive frequencies are paired and offset by 45 MHz.

Frequency assignment, signaling protocols and general control of all terminals will be provided by the DAMA control system which will utilize digital data transmission at a rate of 2.4 kbps.

The major blocks of an ACSSB mobile terminal are [4]:

a) The transmitter
b) The receiver
c) The frequency synthesizer
d) The reference frequency recovery circuit
e) The DMSK modem (for DAMA communications)
f) The ACSSB modem
g) The power supply
h) The control unit
i) The logic unit
j) RF connecting cables
k) The antenna network
l) The T/R switch

Figure 4.1 shows the block diagram of an ACSSB half-duplex terminal. The function of each block is briefly discussed below.

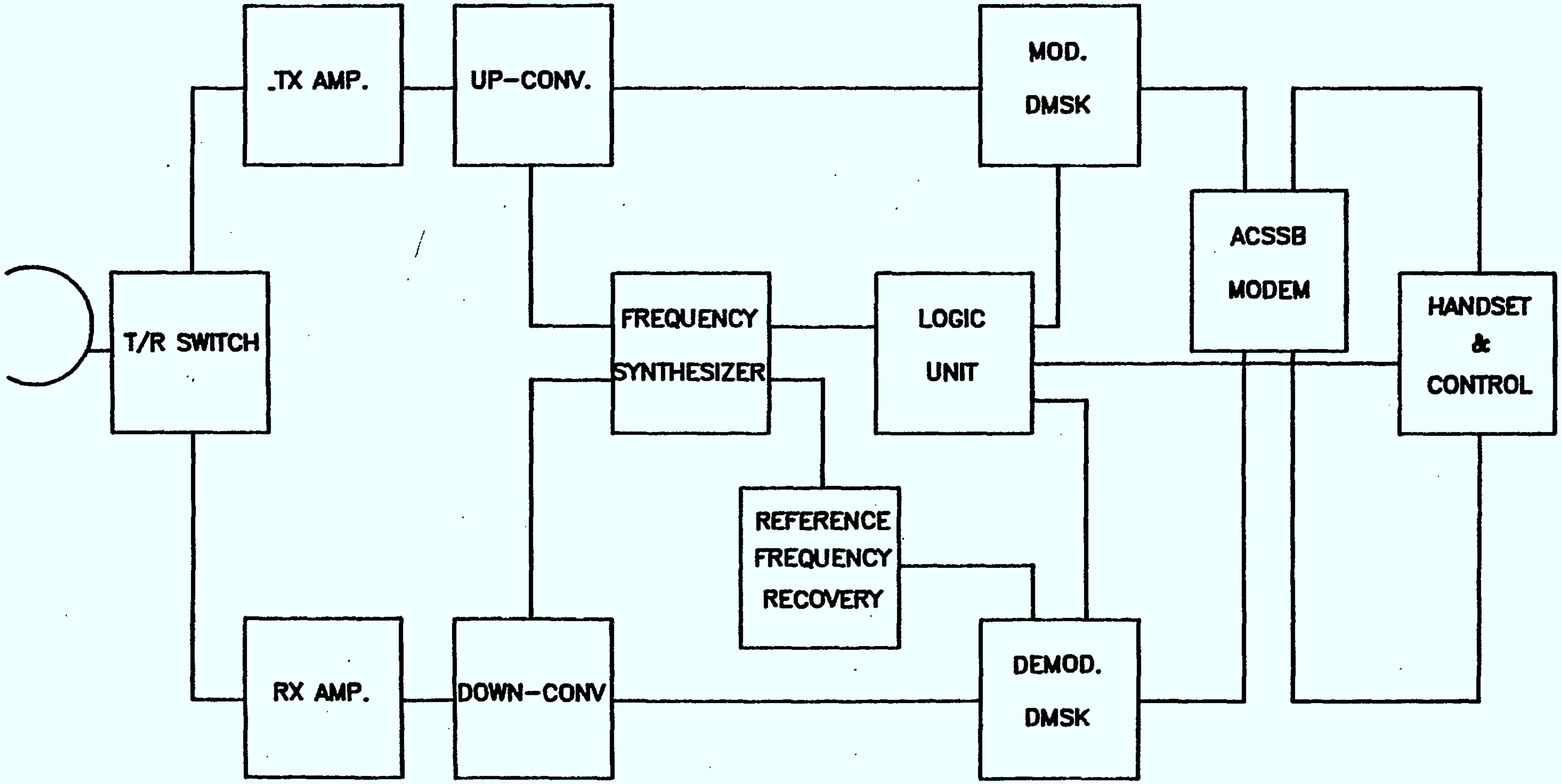

Figure 4.1 : Block Diagram of an ACSSB Terminal

a-b) Transceiver

The transceiver operates in the assigned UHF band in half-duplex mode (push-to-talk). The transmitted output power may vary depending on the type of operations (voice or data transmission) and the antenna used.

c) Frequency Synthesizer

The frequency synthesizer is controlled by the logic unit. When DAMA assigns transmit and receive frequencies to the terminal, the logic unit instructs the frequency synthesizer to provide those frequencies to the transmit and receive chain mixers. The transceiver is capable of tuning over the 4 MHz transmit and receive bands at 5 kHz spacing. The reference frequency for the synthesizer will be derived from the reference recovery circuit, and that reference pilot is distributed via the satellite on a DAMA channel to correct the local oscillator's frequency drifts. This method of frequency correction is necessary to maintain the stability of the oscillator for the duration of a call which is foreseen for MSAT.

d) Reference Frequency Recovery Circuit

Due to the effect of Doppler shift, phase noise in the receiver, and the narrow channel spacing of 5 kHz in the MSAT network, it is necessary to achieve and maintain a very precise control of the oscillator reference frequency for the synthesizer. To accomplish this, a reference signal will be extracted from the modulated data on the DAMA control channel, and this will establish the terminal reference oscillator frequency to the necessary accuracy. During a call, when the DAMA control channel is not available at the terminal, the recovery circuit will continue to provide the necessary frequency stability of $\pm$ 0.1 ppm by maintaining corrected oscillator frequencies over the duration of the call.

e) DMSK Modem

The terminal is assumed to include a DMSK modem operating at a rate of 2.4 kbps. This modem, with an AFC tracking loop, is used to communicate with the DAMA processor in order to establish a communication link.

f) ACSSB Modem

The characteristics of the modem shall be optimized to operate in the fading environment. The TTIB system appears to be the most suitable scheme for MSAT.

g) The Power Supply

The terminal will be powered from the vehicle's main power supply. Thus, the design of the terminal should be compatible with the available resources on a small vehicle.

h) Control Unit

The control unit will provide the user with an interface with the terminal to set up a communication link. The unit will include a handset with push-to-talk button for half-duplex operation (see Appendix C for control unit operation).

i) Logic Unit

This is a microprocessor-based unit which will provide all internal functions necessary for the terminal to respond to DAMA messages; it will also provide data packets for transmission via the DAMA circuit to the DAMA controller. It will allow serial transmission of data in a format which is compatible with the DMSK modem. Frequency selection, transmit power level, and channel request are all functions performed by the logic unit.

j-k) Antenna Network

An important element of the terminals is the antenna network. Omni-directional antennas (4 dBic) as well as more directive antennas with 8 dBic gain will be used in the system. RF connecting cables will be used to connect the antenna termination to the RF modules (LNA, HPA, T/R switch.)

1) T/R Switch

This switch is used in half duplex terminals. It is controlled by the logic unit which commands the terminal into either the transmitting or receiving mode.

4.2.1 Design Life

The equipment should be designed using components and construction methods which are predicted to be available over the next decade. The terminal shall be constructed using materials which do not require special selection. The terminal design life should be at least ten years.

4.2.2. Maintainability

The equipment must be suitable for maintenance by qualified people trained on the equipment. The manufacturer(s) may take the responsibility of repairing faulty modules.

Ease of maintenance and minimizing troubleshooting and repair time should be a prime consideration in the design of this terminal. Each section of the terminal should be built on a separate printed circuit board (PCB) to facilitate replacement and, therefore, to have a minimum repair time.

4.2.3 Reliability

The mean-time-between-failure (MTBF) of the terminal when subjected to normal operating environments should be at least 10,000 hours. Conformance with this requirement may be demonstrated by the testing of a prototype model and by calculating the MTBF.

4.2.4 Physical Characteristics

The land mobile version of the terminal should be suitably packaged to permit installation and operation in a variety of vehicle types. For this reason, the equipment should be miniaturized to the maximum extent practical, taking into consideration cost, performance, reliability, and maintainability. Support and mounting arrangements should be suitably universal and flexible in order to permit installation and interfacing with various vehicles.

Typical dimensions and weight of a terminal are expected to be:

67 mm H x 240 mm W x 280 mm L,
weight 5.3 kg.

4.2.5 Environmental Conditions

The terminal should operate and meet certain minimum performance standards under various environmental conditions. A summary of these environmental conditions is:

Temperature: -40°C to 60°C (storage and operation)

Relative Humidity: 90% minimum

The terminal should be designed to operate under various driving conditions without any effect on its functioning.

4.3 Transmitter Performance

This section provides the expected specifications for the following transmitter parameters:

- Transmitter EIRP
- Transmitter power output
- Transmitter peak radiated power
- Transmitter third order intercept
- Transmitter pilot power level
- Transmitter SSB suppressed carrier
- Transmitter frequency response
- Transmitter adjacent channel spillover
- Carrier inhibit
- Spurious emissions
- Transmitter 1 dB compression point
- Transmitter activation and deactivation time
- Transmitter frequency accuracy

As it was mentioned earlier, the specifications given here are based on the current understanding of the system design and have not been finalized yet. Thus, changes in Telesat's system design on related technological advances could result in a revision of these specifications.

4.3.1 Transmitter EIRP

The maximum effective isotropic average radiated power from the antenna for a UHF terminal is expected to be 11.0 dBW, when no uplink power control is implemented.

4.3.2 Transmitter Power Output

The transmitter power output is the RF power dissipated in the standard output termination when operated under the intermittent rated duty cycle specified by the manufacturer [6]. The average output power assuming an 8 dBic antenna gain will be 3 dBW during speech. However, it is desirable to have a range of average output power from 0 dBW to 5 dBW.

4.3.3 Transmitted Peak Radiated Power

The peak-to-average power ratio is 10 dB assuming that the power will not exceed the specified peak for 1% of the time. Thus the peak power should be in the range of 10 dBW to 15 dBW.

4.3.4 Transmitter Third Order Intercept

The transmitter third order intercept is defined as the intersection of the asymptotes of the characteristics of the Input/Output of the HPA and the third order intermodulation power output. The maximum intermodulation product of two tones at 1.0 kHz and 1.7 kHz respectively both at the peak average power level (7.0 dB above average level) should be at least 30 dB below the audio peak average level. The corresponding third order intercept will be 22 dBW (see Appendix A).

4.3.5 Transmitter Pilot Power Level

The TTIB pilot could be at the average speech power level (0 dBW), and in that case, it should be voice activated. The pilot will be placed at the centre of the MSAT channel. Since the transmitter average power output is 3 dBW during speech, the pilot power level will be 0 dBW.

4.3.6 Transmitter SSB Suppressed Carrier

The SSB carrier level should be at least 30 dB below the average power level.

4.3.7 Transmitter Frequency Response

This is a measurement of output signal level versus frequency with constant audio input level. The transmitter output will be flat within ±1 dB in the 5 kHz bandwidth with an audio input signal at baseband. The 3 dB bandwidth of the transmitter will be 4 MHz (see Appendix A).

4.3.8 Transmitter Adjacent Channel Spillover

The power measured within the 5 kHz bandwidth of the adjacent channel should be attenuated by 30 dB below the average power. The RF signal will be attenuated by 35 dB with respect to the average power at a 5 kHz offset from the nominal transmit channel frequency (see Appendix A).

4.3.9 Carrier Inhibit

This is the reduction in transmitter output power when the output power is inhibited because the frequency synthesizer is in an out-of-lock condition or the reference frequency recovery circuit is inoperative. The transmitter should be switched off under such conditions.

4.3.10 Spurious Emissions

All spurious emissions (excluding harmonics of the signal frequency) will be at least 50 dB below the average power within a frequency band $\pm$500 kHz centred around the pilot. The spurious signal levels will decrease monotonically to 70 dB below the average power at frequencies $\pm$1 MHz or greater. The RF harmonic attenuation will be at least 70 dB below the rated power.

4.3.11 Transmitter 1 dB Compression Point

Normally, the 1 dB compression point of an SSPA is about 1 to 2 dB below the peak level [8]. The 1 dB compression point of the transmitter is the output signal level of the HPA which has fallen by 1 dB from the asymptote of the linear portion of the input/output signal curve. The transmitter 1 dB compression point should be 12 dBW.

4.3.12 Transmitter Activation and Deactivation Time

The transmitter pilot tone will be voice activated. The transmitter will be activated within 5 msec after the reception of the keying signal. The pilot level will be reduced to a very low level below its steady state power within 5 msec of the removal of the keying signal. The level measured by Adga was 12 dB below its steady state level. However, it is felt that about 30 dB will be sufficient.

4.3.13 Transmitter Frequency Accuracy

The transmitter frequency accuracy with AFC will be within $\pm$0.1 ppm of the broadcast reference frequency over the specified temperature range in which the terminal will be operative.

4.4 Receiver Performance

4.4.1 Receiver Noise Figure

The receiver noise figure is the ratio of actual signal-to-noise ratio at the input of the receiver, to the signal-to-noise ratio at the output of the receiver. The receiver's noise figure should not exceed 2.0 dB (see Appendix A).

4.4.2 Receiver Sensitivity

This is the minimum signal level which can be detected in the presence of noise. The minimum signal level will be about -136 dBm.

4.4.3 Receiver Third Order Intercept

The third order intercept point is defined as the point at which the asymptotes of the input/output transfer curve and the third order intermodulation products curve intersect. The third order intercept point is determined at the output of the low noise amplifier, and should be at least 42.5 dBm for worst case interference. However, a third order intercept LNA of 20 dBm is acceptable in the case of a more relaxed interference requirement (see Appendix A).

4.4.4 Receiver Audio Frequency Response

This is a measurement of audio output signal level versus frequency with constant RF input signal level. The audio output from the receiver will be flat within $\pm$ 1 dB over the band 300 to 3000 Hz. The response above 3000 Hz will roll-off monotonically and will be no higher than -25 dB at 4000 Hz (see Appendix A).

4.4.5 Receiver Audio Output Volume Control [2]

The audio output of the receiver will be adjustable from the front panel over a range of 30 dB.

4.4.6 Receiver Squelch

The minimum signal level at the channel frequency which will open the receiver squelch will be -136 dBm. The squelch attack time will be no greater than 5 msec and the decay time no greater than 5 msec.

4.5 Frequency Synthesizer Performance

The following items are of particular concern for the frequency synthesizer:

- Frequency selection
- Spurious emissions
- Phase noise
- Switching speed

4.5.1 Frequency Selection

Frequency selection is the ability of the synthesizer to provide upon a DAMA command each of the required Local Oscillator (LO) frequencies, corresponding to each discrete channel at 5 kHz spacing, for both the transmitter and the receiver.

The synthesizer should be able to provide 800 frequencies at 5 kHz increments. Transmitter frequencies will be from 821.0 MHz to 824.995 MHz while receiver frequencies are from 866.0 MHz to 869.995 MHz. Each transmit and receive channel pair will be separated by 45 MHz.

4.5.2 Spurious Emissions

Spurious emissions are emissions at any frequency away from the selected synthesizer frequency. These include harmonic emissions, parasitic emissions, and exclude noise-like emissions due to phase modulation by noise. The spurious emissions should be less than 70 dB below the main carrier frequency level [2].

4.5.3 Phase Noise

This is defined as the phase instability of the oscillator measured in the frequency domain. The SSB phase noise should decay at least to -30 dBC at 10 Hz away from the carrier and to -70 dBC at 1 kHz offset (see Figure 4.2) [2].

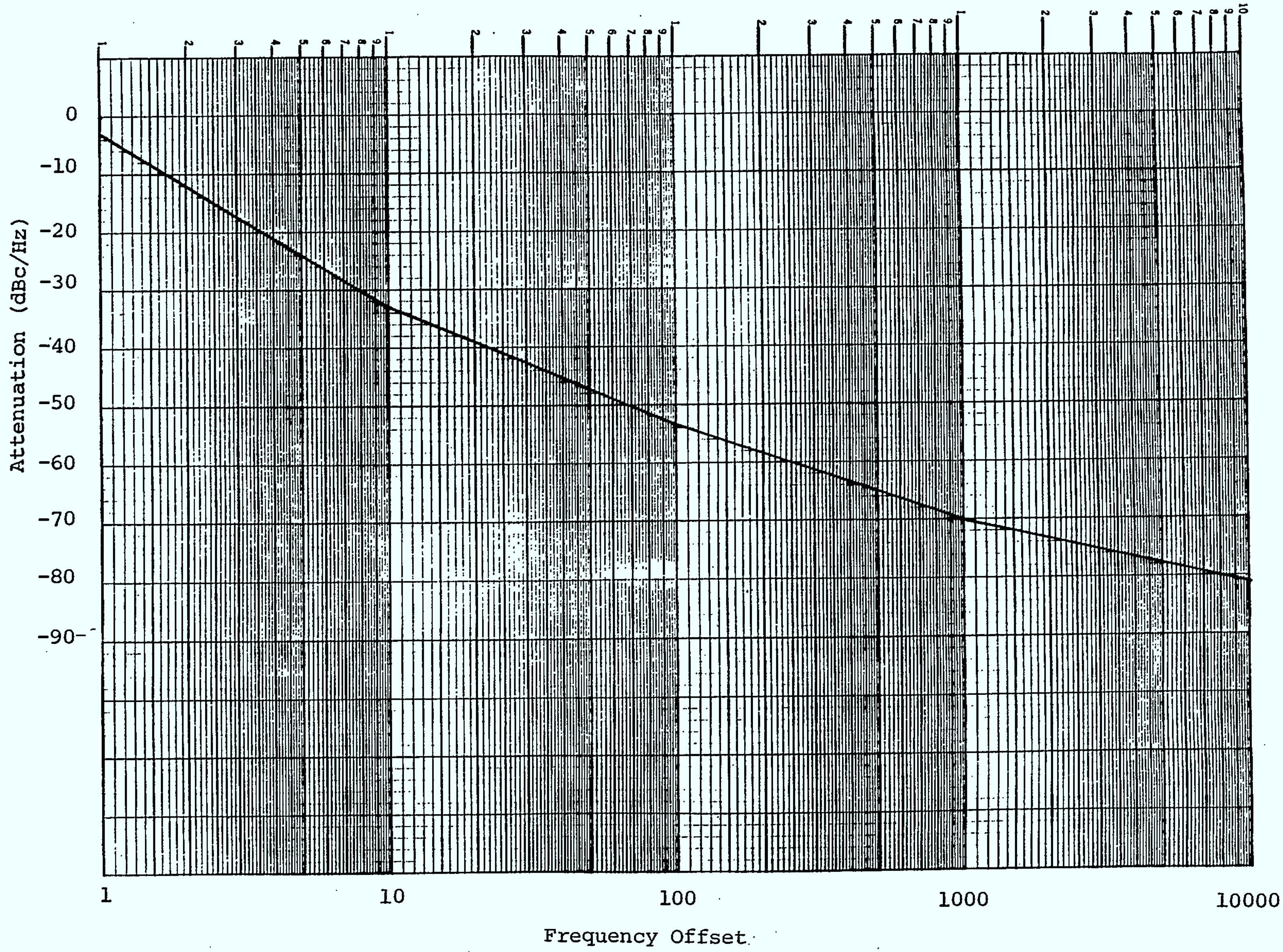

Figure 4.2- Frequency Synthesizer Phase Noise Characteristics

4.5.4 Switching Speed

This is the time delay between the instant a frequency selection is required from the logic unit and the instant that that frequency is present at the synthesizer output. The synthesizer switching speed and the time which the oscillators need to lock on the corrected frequency will be less than the one-way trip delay between the instant the terminal sends an acknowledgement to DAMA and the time DAMA receives it. The synthesizer switching speed should be 150 to 160 msec (see Appendix A).

4.6 Power Supply Performance

The UHF mobile terminal will be powered by the standard vehicle battery.

• Nominal input voltage range:	11.0 V - 15.0 V
• Regulated output voltage:	13.6 V
• Maximum ripple, regulated supply:	20 mv rms.

4.7 Control Unit Performance

The control unit is the device which provides the user with an interface with the terminal and through it the user is capable of setting up a communication link. The unit will consist of a control panel and a handset.

For the land mobile version of the terminal the control unit will most probably be remote from the transceiver and, thus, the user should be able to access the unit with convenience.

The control unit will interface with the DAMA logic unit using digital signals.

4.7.1 Control Unit Features

Many features are required in the control unit to indicate the state of operation of the mobile terminal. Those features are:

- Keypad
- Numeric display
- On/Off switch
- Hook switch
- Ringer
- Speaker
- Push-to-talk switch

Several indicators will also be included. Those are:

- Transmit/Receive
- Power on/off
- On-hook/off-hook
- Abort
- Log-on/off
- State of DAMA signal strength

4.7.2 Control Unit Operation

After the log-on operation, the operator enters the ID number of the called party when attempting a call. The numeric display is used to display ID numbers punched in through the keyboard and to display received call ID numbers.

The on/off switch should turn the power on/off and once turned on, the processor will not transmit until the correct password has been entered. The password is necessary to prevent misuse or access to the system by stolen units.

At the start of a conversation, an off-hook message is transmitted to the DAMA processor and the off-hook function indicator is set. At the end of the conversation, the DAMA processor is notified and the LED indicator is reset.

The ringer is used to inform the user of a busy, available, or unavailable called party. In the voice mode, the push-to-talk switch is used to control the operation of the T/R switch which should respond in less than 150 msec. However, during DAMA and logic unit communication, the push-to-talk switch is disabled.

The following table shows the preferred modes of indicator illumination :

Function Indicator	Illuminated
Transmit/Receive	Transmit
Power on/off	Power on
On-hook/off-hook	Off-hook
Abort	Call aborted
Log-on/off	Log-on

4.8 Logic Unit Performance

The logic unit will control the operation of the radio. Furthermore, it will respond to commands from the DAMA processor. The human interface with the unit will be via the control unit. The logic unit will be microprocessor-based with a stored program. The desirable types of command received from the DAMA are:

- Assigned voice channel frequencies
- Transmit power level of the mobile
- Reference frequency used for correction of L.O. drifts
- Reference time to be used in request channel synchronization.
- Signaling channel frequencies for Dynamic signaling channel assignment.

The logic unit will have the following interface signals as a minimum.

Modem:

- Tx data at 2.4 kbps
- Tx clock at 2.4 kHz
- Rx data at 2.4 kbps
- Rx clock at 2.4 kHz
- Request to send
- Clear to send
- Reset

Frequency Synthesizer:

- Frequency select
- Lock indicator

RF Unit:

- Power control (voice/data)
- Power on/off
- Power detection

Control Unit:

- Data keyboard
- Switches
- Indicators (visual and audible)

4.9 DMSK Modem Performance

The DMSK modem, being the most probable choice for data transmission (see chapter III), consists of a modulator and a demodulator, and it interfaces with the RF circuitry and the logic unit. The modem's characteristics are defined as [3]:

The Modulator

Nominal Bit Rate	2.4 kbps
Channel Bandwidth	5 kHz
Modulation technique	Differential Minimum shift keying
Delay from input to output	0.4 ms maximum

The Demodulator

Nominal Bit Rate	2.4 kbps
Demodulation Technique	Differential Detection
Error Correction	Single Error Correction (15, 11) BCH Code
Maximum Frequency Offset	$\pm$440 Hz (equipped with AFC)
Bit Error Rate	See Figure 4.3
Automatic Gain Control (AGC)	Optional
Delay From Input to Output	4.0 ms

Baseband Interface

Tx Data	2.4 kbps
Tx Clock	2.4 kHz
Rx Data	2.4 kbps
Rx Clock	2.4 kHz
Request to Send	Optional
Clear to Send	Optional

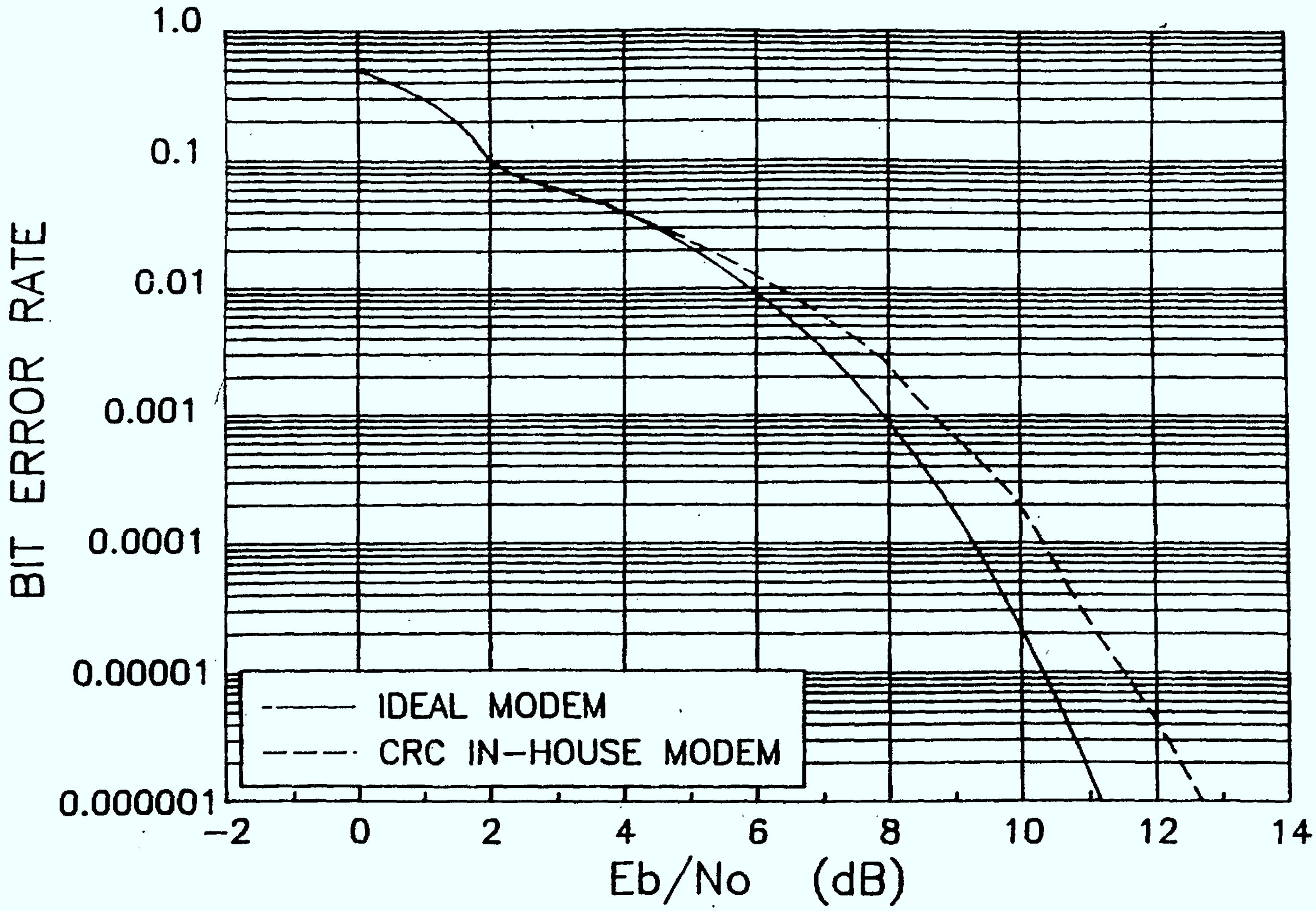

Figure 4.3: Bit Error Rate Performance of DMSK

Prior to transmitting data, the DMSK modulator will transmit a sequence of bits, called the preamble which will be necessary for the demodulator to perform automatic frequency compensation and bit timing recovery on the received signals. Thus, a message would contain the bit sequence including the preamble and the source data. The preamble sequence will be 32 bits long for AFC purpose and 24 bits long for bit timing recovery. Thus, the total length of the preamble is 56 bits.

4.9.1 Modulation Index

The modulation index of a DMSK signal is the ratio of peak-to-peak deviation to the data rate. The peak-to-peak deviation will be 1.2 kHz yielding a modulation index of 0.5.

4.9.2 Out-of-Band Spurious Emissions

These are emissions at any frequency outside the 5 kHz bandwidth, including carrier harmonics and intermodulation products. The out-of-band emissions of the CRC DMSK modem are 40 dB below the carrier level. This value is also suggested here.

4.9.3 Adjacent Channel Power

This is the part of the power output of the modulator which falls within a 4 kHz bandwidth, centred at 5 kHz away from the carrier frequency. This power for the CRC modem is 40 dB relative to the wanted channel power level. This value of adjacent channel power is also recommended here.

4.9.4 Bit Error Rate of the Demodulated Signal

This is the probability of error measured with a baseband receiver. Figure 4.3 compares the probability of error vs. E_b/N_o of an ideal modem with that of the CRC in-house design. Note that the difference between performance of the two modems at BER=10^{-4} is about 1 dB.

4.10 ACSSB Modem Performance

As it was mentioned previously, TTIB seems the most favourable technique to operate in MSAT's fading and shadowing environment. The desired characteristics of the interfacing sections are given below.

Audio Interface

The audio input and output will be via a telephone-type handset. The type of microphone and earphone will be recommended by the manufacturer.

IF Interface

It is recommended to be the same as the CRC design:

Frequency:	4.8 kHz ± 10 Hz
Input Level:	-10 dBm to -23 dBm measured at the equivalent average speech power
Output Level:	0 dBm ± 1 dB
Output Impedance:	50 ohms

Baseband to Baseband Performance

C/No	Speech Quality
50 dB-Hz	Good Quality
45 dB-Hz	Fair Quality
40 dB-Hz	Threshold of Intelligibility

4.10.1 Automatic Level Control (ALC)

This is a measure of the ability of the equipment to ensure that the signal output does not exceed the peak power. By varying the input signal by 10 dB, the output power should not change more than 1 dB.

4.10.2 Companding Ratio

Compression is used to reduce the dynamic range of the speech signal, allowing it to be transmitted undistorted through a noisy channel. Companding is performed in the transmitter by compression of speech amplitude. The reverse operation is performed in the receiver. The compression and expansion ratios are complementary. The companding ratio recommended for SSB radios is 3:1.

4.10.3 Automatic Frequency Control (AFC)

The AFC limits are the upper and lower frequency limits relative to the pilot frequency within which the automatic frequency control is operative. For an ACSSB radio using the TTIB technique, the AFC limits will be $\pm$400 Hz.

4.10.4 Automatic Gain Control (AGC)

The AGC characteristic is the change in audio output power as the RF input level is varied over a specific range. For the ACSSB receivers, varying the RF input signal by 25 dB (from -140 dBm to -115 dBm), the audio output variation should not be more than 3 dB.

4.10.5 De-Emphasis

The de-emphasis operation is required for noise shaping or suppression in the received audio signal. Since high frequency components of the speech have less energy than the low frequency components, de-emphasis is effective in reducing the high frequency noise density. Hence, an advantage in S/N is accomplished. A 12 dB/octave de-emphasis, is recommended to be included in the TTIB processing [1].

4.11 RF Connecting Cables

Cables are used to establish a connection between the RF section of the terminal and the antenna. The following parameters define the characteristics of the RF connecting cables:

- Impedance:	50 ohms
- Length:	3 meters maximum
- Maximum Attenuation at 850 MHz including RF connectors:	1.5 dB

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CHAPTER V - COMPATIBILITY OF MSAT WITH TERRESTRIAL SYSTEMS

5.1 Introduction

In this chapter, the implications of modifying the conventional terrestrial terminal designs to allow for operation in the MSAT system are examined.

Modifications of both FM and ACSSB terminals, or any suitable spin-offs from these technologies, are addressed in this chapter. However, each system will be considered separately, since their requirements are different.

5.2 Comparison of MSAT ACSSB and FM Terrestrial Systems

Although processing requirements for FM are simple to implement, an FM radio needs many modifications in order to be compatible with the MSAT system. The critical performance requirements for operation within MSAT are different from those of available land mobile radios, such as:

- frequency coverage,
- channel spacing,
- channel bandwidth,
- receiver sensitivity,
- intermodulation response,
- transmitter power control,
- modulation technique.

The following briefly addresses the critical differences between the two types of terminal.

5.2.1 Signal Processing

The baseband signal processing requirements in ACSSB using TTIB, are different from those required for conventional FM terrestrial mobiles. For MSAT terminals in transmission mode, various signal processing functions are performed on the audio signal to generate a transparent tone-in-band signal. These functions are: bandsplitting, ALC, compression, pre-emphasis, and pilots insertion. In the receiver, first Feed-Forward Signal Regeneration (FFSR) is performed to correct for phase fluctuations and fading, then expansion, de-emphasis, and subband combining operations are applied.

Therefore, the baseband processing of ACSSB radios for use in MSAT service is significantly more complex than that of FM terrestrial radios.

5.2.2 Frequency Coverage

The major constraint in obtaining the desired frequency coverage would be due to the RF filtering requirements. The frequency bands allocated to FM terrestrial mobiles are 806 MHz to 821 MHz for transmit, while the receive band is from 851 MHz to 866 MHz. It is seen that a new RF section would be used for MSAT terminals.

5.2.3 Channel Spacing

The FM systems operate on the basis of 30 kHz channel spacing which is six times greater than the proposed 5 kHz spacing for MSAT. Therefore, a redesigning of the frequency synthesizer is a must for MSAT terminals.

5.2.4 Channel Bandwidth

The different bandwidth and modulation modes affect the transmitter design. The MSAT ACSSB terminal requires a narrow IF filter with 3 dB bandwidth of 4 kHz. Wider IF filters are necessary in the FM mode. The channel bandwidth in the proposed MSAT ACSSB system is at most 3.9 kHz while it is 27 kHz for Advanced Mobile Phone Systems (AMPS). It is clear that the IF section design for the two terminals are very different from each other and a new design for MSAT terminals should be used.

5.2.5 Receiver Sensitivity

MSAT terminals require very low noise amplifiers which would be saturated by the relatively high power of terrestrial stations. The noise floor of UHF MSAT receivers is -165.6 dBW/5kHz while the noise floor of the conventional 800 MHz terrestrial radios is -151 dBW/30 kHz. Therefore, a very sensitive front end receiver is required for MSAT terminals.

5.2.6 Frequency Stability

The required frequency stability of an MSAT terminal is higher than that of the terrestrial terminals. The required L.O. frequency stability of MSAT mobile terminals is expected to be $\pm$ 1 ppm to 2 ppm whereas that of the conventional FM systems is $\pm$ 2.5 ppm. Note that the overall frequency stability of MSAT terminals after pilot acquisition is $\pm$ 0.1 ppm.

5.2.7 Audio/Baseband Frequency Response

The baseband IF filter used to recover the audio signal would have the same characteristic for both types of service. The 3 dB bandwidth of FM terrestrial receivers and that of the MSAT receivers is 2700 Hz (300-3000 Hz).

5.2.8 Transmitter Power Output

The High Power Amplifiers (HPA) used in MSAT and FM terminals have different characteristics. The ACSSB transmitter will be designed for linear operation at an average power varying from 0 to 5 dBW and at a peak power 10 dB higher than the average level. In the FM transmitter, the average power is equal to the peak power of 10 dBW since it is a constant envelope modulation. Since the efficiency of a linear PA (class A) is approximately 20% compared with 60% for class C FM transmitters, the required heat sink for MSAT terminals will be comparable to FM transmitters.

TABLE 5.1

Performance Requirements for FM and MSAT

Item	FM (cellular)	ACSSB (MSAT)
Frequency coverage	Transmit:806-821 MHz Receive:851-866 MHz	Transmit:821-825 MHz Receive:866-870 MHz
Channel spacing	25 kHz for radio 30 kHz for mobile phones	5 kHz
Channel bandwidth	27 kHz for mobile phones 16 kHz for radio	3.9 kHz
Frequency Stability of L.O.	± 2.5 ppm	± 1 to 2 ppm
Receiver Sensitivity	-109 dBm	-136.0 dBm
Audio freq. response	300 to 3000 Hz	300 to 3000 Hz
Transmitter power output	10 to 13 dBW	0-5 dBW average
Transmitter modulation	FM with ± 5 kHz peak deviation	ACSSB (TTIB)
Mode	(MRS) Half duplex for radio (MTS) Full duplex for telephone	(MRS) half duplex
Transmitter HPA	Class C	Class A
Filters requirement	70 dB at 30 kHz adjacent channel isolation	35 dB at 5 kHz adjacent channel isolation

5.2.9 RF and IF Filtering and Interference Suppression

With the proposed MSAT UHF spectrum allocation, a difficult parameter affecting the receiver cost could be interference. With the projected distribution of mobiles, base stations, and TV transmitters by the year 2000, MSAT receivers might require substantial RF filtering together with high third order intercept LNAs and mixers if performance degradation due to interference is not to affect the overall system availability. A more relaxed filtering is acceptable in FM receivers.

In a terrestrial mobile system, 70 dB of adjacent channel isolation at 30 kHz spacing is required, while, in the MSAT system, the IF signal will be attenuated by 35 dB at 5 kHz away from the center of the channel. This may impose severe constraints on the IF filtering requirements of the MSAT terminals if they are to operate in proximity of the terrestrial interferers.

5.2.10 Summary

From the above discussions it is clear that only few spin-offs could exist from the conventional FM terrestrial mobile technology.

Table 5.1 summarizes the performance requirements of FM terrestrial terminals and MSAT ACSSB terminals. It is seen that the performance requirements for MSAT terminals differ substantially from those of available FM land mobile equipment. In particular, the areas of concern are:

- Receiver sensitivity
- Frequency coverage
- Channel bandwidth
- Modulation scheme
- Signal processing
- Frequency selection

ADGA has reported that the interference suppression requirement at UHF could significantly affect the design and cost of MSAT receivers. However, it is believed that a relaxed interference suppression could be used for MSAT mobiles operating in rural areas (see Appendix A).

5.3 Comparison of MSAT Terminals with Terrestrial ACSSB

The ACSSB terminals for MSAT will employ the TTIB technique and are thus not compatible with the ACSSB (TAB) terminals which are currently subject to DOC and FCC approval.

At present, it appears that few terrestrial mobile receivers conforming to the DOC and FCC standards for operation in the 800 MHz band are in the implementation process. The closest available terminal to the desired characteristics is the one developed for use in the airfone system by E.F. Johnson in the Unites States. However, this airfone uses compatible single sideband (CSSB) where the pilot reference is transmitted at a high level. Therefore, it is useful to compare the requirements of MSAT and terrestrial ACSSB systems in order to draw some conclusions on possible spin-offs from the terrestrial ACSSB terminals for MSAT system.

Table 5.2 summarizes the performance requirements of MSAT and terrestrial ACSSB. It appears that most of the specifications in the two systems are identical or about the same except for the receiver sensitivity, the intermodulation response, and the L.O.'s frequency stability. However, there are some hidden differences which are explained below.

5.3.1 Channel Bandwidth

Although the two systems appear compatible in channel bandwidth, the placement of the pilot tone at the band edge (TAB), as used in the terrestrial ACSSB, may result in performance degradation unless the amplitude and group delay characteristics are carefully controlled. This is a drawback in the terrestrial system which needs a more precise IF filter in order not to distort the pilot. This requirement is relaxed for MSAT terminals using the TTIB technique.

TABLE 5.2

Performance Requirements for ACSSB and MSAT

Item	ACSSB (terrestrial)	ACSSB (MSAT)
Frequency coverage	Transmit:806-821 MHz Receive:851-866 MHz	Transmit:821-825 MHz Receive:866-870 MHz
Channel spacing	5 kHz	5 kHz
Channel bandwidth	3.4 kHz	3.9 kHz
Frequency Stability after pilot acquisition	± 0.38 ppm	± 0.1 ppm
Receiver Sensitivity	-110 dBm	-136.0 dBm
Audio freq. response	300 to 2500 Hz	300 to 3000 Hz
Intermodulation response	-60 dB	-90 dB
Transmitter power output	14 dBW average	0-5 dBW average
Transmitter modulation	ACSSB (TAB)	ACSSB (TTIB)
Mode	(MRS) Half duplex	(MRS) half duplex
Transmitter HPA	Class AB	Class A
IF Filters requirement	35 dB at 5 kHz adjacent channel isolation	35 dB at 5 kHz adjacent channel isolation
Receiver AGC	10 dB output variation over 80 dB input range	3 dB output variation for 30 dB input range

5.3.2. Receiver Sensitivity

For MSAT, an overall receiver noise figure of 2 dB is required, whereas the terrestrial systems do not need such a stringent requirement. The terrestrial terminal's receiver sensitivity of -110 dBm is inadequate for use in MSAT where the required receiver sensitivity is about -136.0 dBm.

5.3.3 Audio Frequency Response

The audio response of the terrestrial ACSSB mode is cut off at the high end by the need for separation from the pilot tone. That is, there is a more stringent requirement on IF filtering of terrestrial ACSSB terminals with TAB if the channel spacing is to be limited to 5 kHz.

5.3.4 Transmitter Power Output

The ACSSB terrestrial terminals apparently require about 25 watts PEP (Appendix E), whereas it is significantly lower for MSAT terminals. Therefore, by using a lower gain power amplifier, MSAT terminals benefit in intermodulation response as well as its associated cost.

5.3.5 Transmitter Modulation

The baseband processing of the two terminals are completely different. In the TAB technique, the pilot is placed above the audio band, hence increasing the adjacent channel interference to the pilot.

5.3.6 Receiver AGC

Fade statistics have shown that very deep fades may occur for a terrestrial mobile. Therefore, the AGC dynamic range of the terrestrial receivers should be large enough to recover the faded audio signal. In the MSAT network, an FFSR loop is used to correct for phase and amplitude fluctuations. FFSR responds well to fast fades since it is a mixing process rather than an AGC-type amplifier. The AGC being a feedback loop, would produce greater delay in the signal path.

5.3.7 Summary

The necessary alterations to the terrestrial ACSSB radio terminals, in order to operate in MSAT system, are mainly confined to the audio and baseband signal processing. In addition, the receiver sensitivities differ substantially in the two systems. The AGC circuit requirements are also different (see Table 5.2).

5.4 Contact With Industry

In the course of this study, several North American mobile terminal manufacturers were contacted in order to obtain information on their current and future products, as well as to explore their plans in the design and manufacturing of MSAT mobile terminals.

A few companies are currently manufacturing amplitude compandored single sideband terminals at VHF and UHF with 5 kHz and 6 kHz channel spacing respectively. Their views about the possibility of converting their products to UHF in order to meet the MSAT requirements were also solicited, despite the fact that both terrestrial and MSAT systems seem to be non-compatible.

The Canadian firms were very reluctant to release information on their current plans; however, they have expressed some interest in participating in MSAT activities.

The organizations contacted are:

- Glenayre Electronics Ltd., Vancouver, B.C.
- Novatel, Calgary, Alberta
- Stephen Engineering Associates (SEA), Seattle, Washington
- E.F. Johnson, Wasica, Minnesota
- Aerotron (Sideband Technology), Raleigh, North Carolina
- Contemporary Communications Company, New Rochelle, N.Y.

Our survey indicated that none of the contacted organizations were in favour of converting an existing terrestrial terminal to an MSAT terminal. They also believed that the minimum performance requirements of a MSAT terminal differ substantially from those of the existing terrestrial terminals (see Appendix E). They expressed the following differences between the two systems:

- Design of a very sensitive receiver for satellite operation
- Sharp RF and IF filters to suppress adjacent band interference
- TTIB with FFSR processing
- A frequency synthesizer with high switching speeds, and accurate frequency lock range.

The record of the conversations with the contacted organizations is given in Appendix B.

CHAPTER VI - THE MSAT L-BAND GROUND TERMINAL

6.1 Introduction

The BNR subjective quality tests which were conducted for 800 MHz UHF fading statistics, have demonstrated that a carrier-to-noise density ratio of about 50 dB-Hz is required for MRS users with ACSSB terminals to achieve a rating of 3 (fair) in a light shadowing/fading environment. The LPC/DMSK terminal at its current level of development was judged by the subjects as poor under all conditions. That was probably due to the unfamiliarity of the subjects with the electronic accent of the LPC vocoder. Therefore, it was judged that ACSSB should be used for voice communication via MSAT, at least for the first generation of MSAT. Meanwhile, further development should be undertaken to improve the performance of LPC.

The objective of this chapter is to predict the voice quality of the ACSSB terminals at L-band, based on interpolation of the UHF test results.

6.2 Voice Quality at L-band

The expected voice quality at L-band depends upon the characteristics of the fading/shadowing environment, which lead to a degradation in the speech channels. Due to the nearly doubling of the frequency of operation from UHF to L-band, the frequency drift, and the Doppler shift, which are dependent upon the frequency of operation, are also nearly doubled. At 850 MHz, the Doppler shift is approximately 80 Hz, while at 1.6 GHz, the Doppler shift becomes about 150 Hz. Assuming that ACSSB with TTIB will again be the modulation scheme used, the width of the notch is nearly doubled. The current CRC design, which is based on the TTIB processing technique, has a notch width of 1200 Hz which could easily accommodate the Doppler shift occurring at L-band. However, a wider notch means a higher level of the TTIB pilot will be required to maintain the same C/N for the pilot.

Figure 6.1 illustrates the light fading/shadowing characteristics for both UHF and L-band. Note that both sets of data are taken from the same area, for the same elevation angle, and the same season. It is seen that 13 dB of fade margin is required for 99% of signal availability for UHF, whereas the corresponding fade margin at L-band is 18 dB.

In addition to deeper fades occurring at L-band, the frequency of occurrence of certain fade levels is higher for L-band, as illustrated in Figures 6.2 and 6.3 [1]. From Figure 6.2, it appears that the frequency of fades at L-band is roughly double that at UHF for a vehicle traveling at about 80 km/hr. From Figure 6.3, the number of fades for a duration of one wavelength is over 2000 for a fade of 9 dB at L-band. For the same duration at UHF and a fade of 7 dB the number of fades is about 1000. Note that from Figure 6.1, a 7 dB fade at UHF corresponds to a 9 dB fade at L-band.

This increase in fade levels at L-band may lead to a poorer voice quality rating by the users if the same fade margin is used in the system design. However, in the absence of shadowing, it is expected that the performance at L-band is comparable to that at UHF.

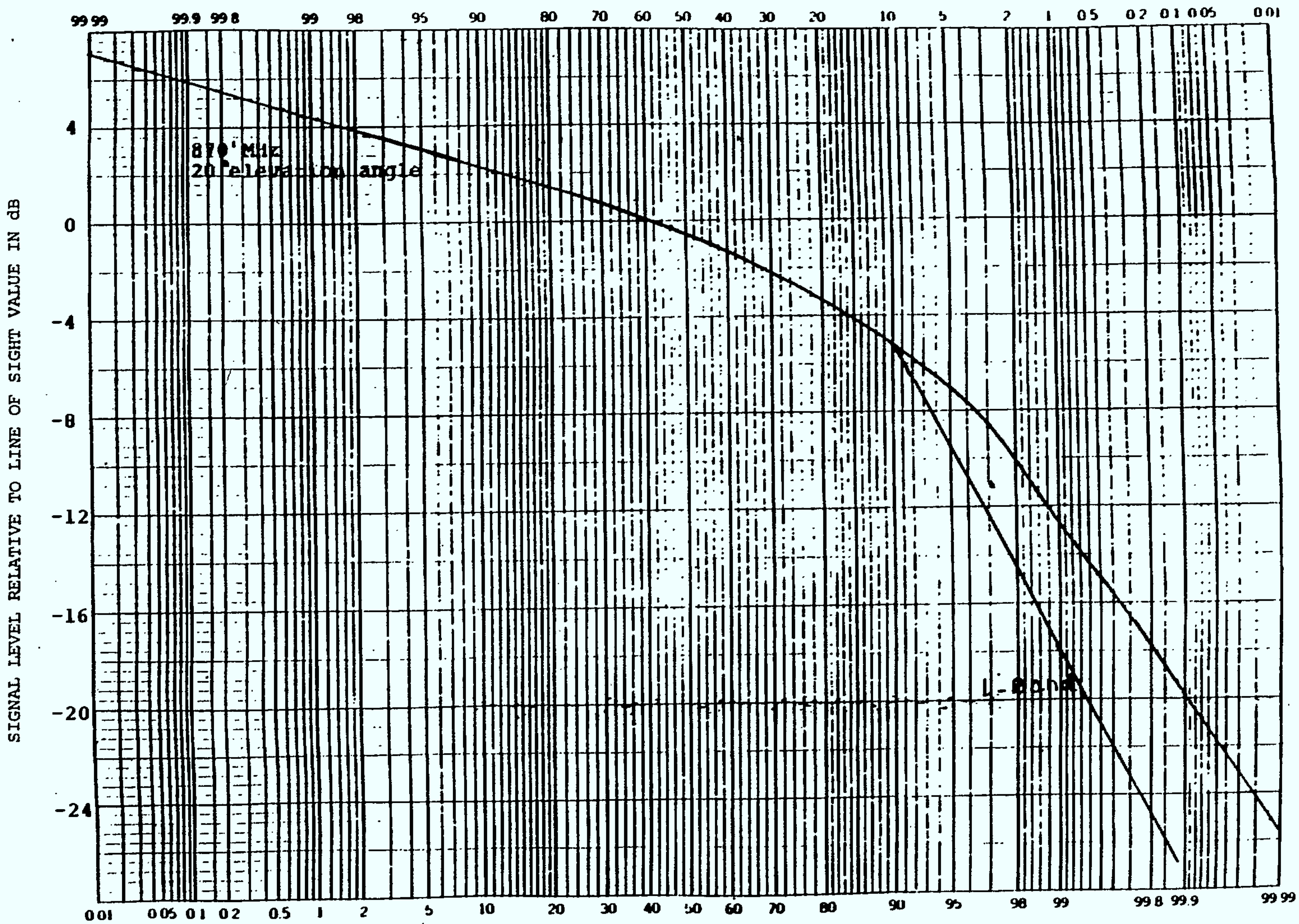
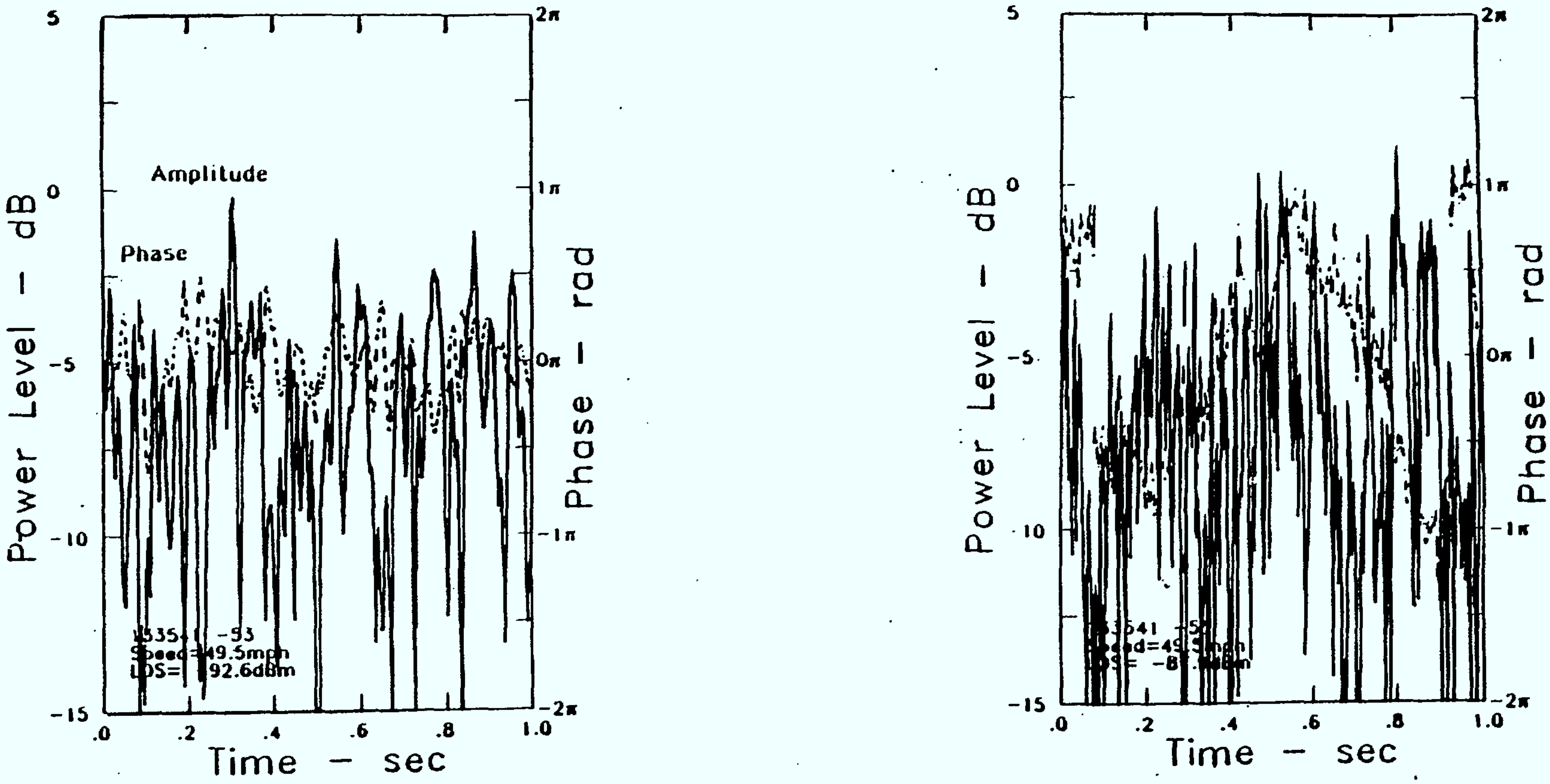


FIGURE 6.1: PERCENTAGE OF THE TIME SIGNAL WAS GREATER THAN ORDINATE



869MHz

1501MHz

FIGURE 6.2: FADES CAUSED BY ROADSIDE TREES ELEVATION ANGLE OF 40 DEGREES AUTOMOBILE SPEED OF 49.5 MPH

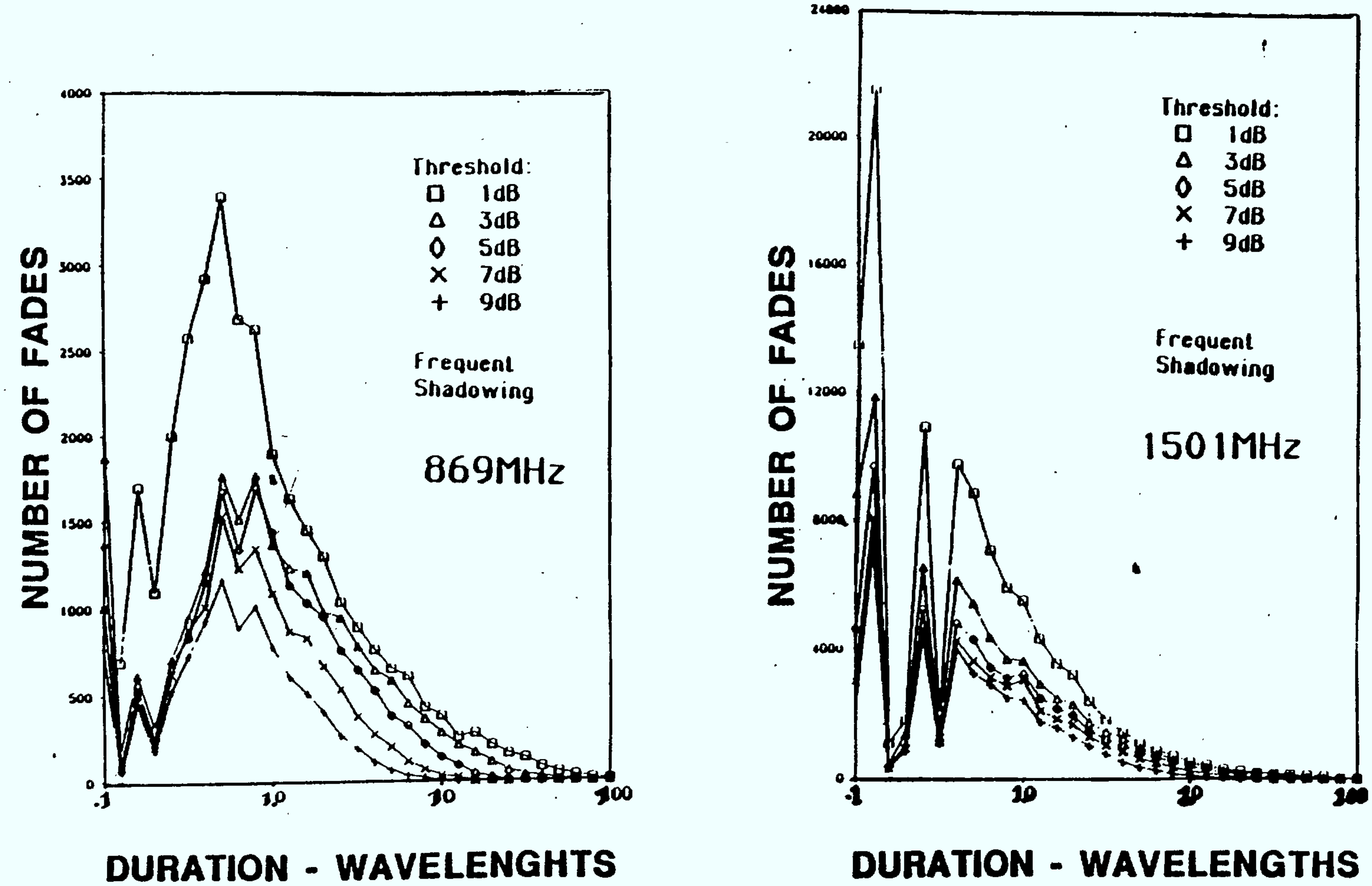


FIGURE 6.3: FADE DURATION HISTOGRAMS FOR FREQUENT SHADOWING BY TREES OVER 55.2 KM TOTAL DISTANCE TRAVELLED

To estimate the subjective voice quality of ACSSB at L-band, we will first compare the fading characteristics of UHF and L-band, as given in Figure 6.4. Specifically, the L-band light shadowing statistics are shown along with the statistics for UHF light and medium fading. From the figure, it is seen that the L-band light fade statistics fall between those of UHF light and medium fade statistics. Therefore, it can reasonably be expected that the voice quality ratings for ACSSB at L-band will lie somewhere between the results for UHF light and medium fadings. The expected ratings for ACSSB at L-band are shown in Figure 6.5. Note that this estimate does not take into account the increased frequency of fade, at L-band. It is seen that, for a light fading condition, an overall C/No of about 53 dB-Hz would be required to accomplish a mean opinion score of 3.0 (fair) at L-band. However, for regions with smaller shadowing losses (e.g. locations with a higher elevation angle or no shadowing losses such as the prairie provinces, the required C/N_o would be less than 53 dB).

6.3 L-Band Terminal Design and Specifications

The design of mobile terminals at L-band will be different from the UHF design, particularly in the antenna and RF sections. With the 5 kHz channel spacing at L-band, the IF and baseband sections are expected to remain unchanged. However, a minor difference in the baseband section could be limited to the adjustment of the width of the notch for the TTIB scheme. This is accomplished by means of a simple software modification. Note that, the CRC design (1.2 kHz notch width accounts for the Doppler spreading and the frequency inaccuracies which are likely to happen at L-band. It is expected that the IF and RF filters design will be simplified at L-band due to the significantly lower interfering signal levels at this band. Therefore, an overall noise figure of 2 dB should technically be easier to achieve at L-band. However, the cost of the L-band components, especially the HPA and the antenna, is expected to be substantially higher than their UHF counterparts. Also, the overall terminal frequency stability of ±0.1 ppm, which should be maintained at L-band, dictates the use of highly stable local oscillators of ± 1 ppm as opposed to the relaxed requirement of ± 2 ppm at UHF.

A detailed investigation of the impact of operating at L-band on the mobile terminal design and the required specifications are not within the scope of the current study. The results of that study will be issued in a separate report in due time.

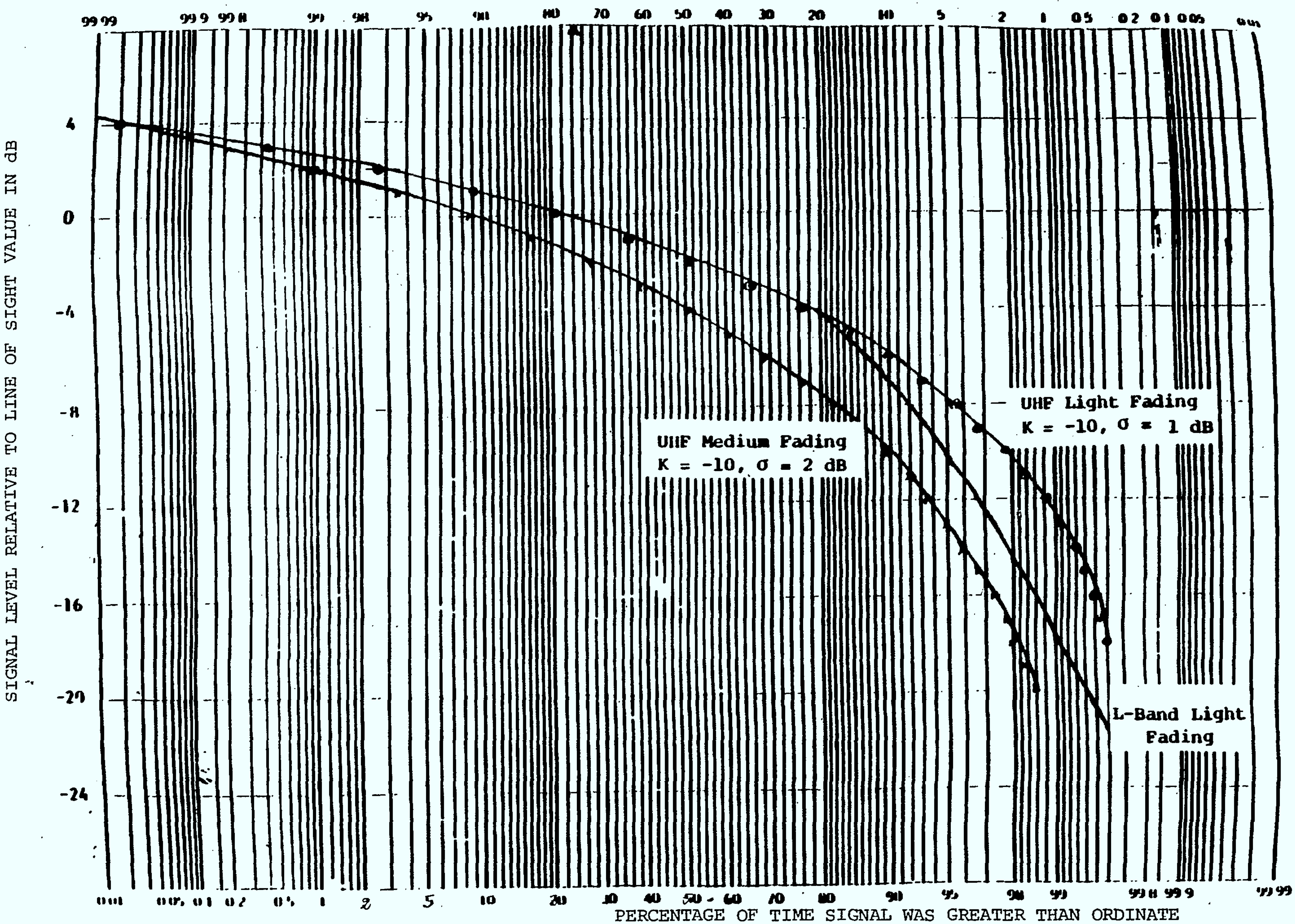


FIGURE 6.4: COMPARISON OF UHF AND L-BAND FADING STATISTICS

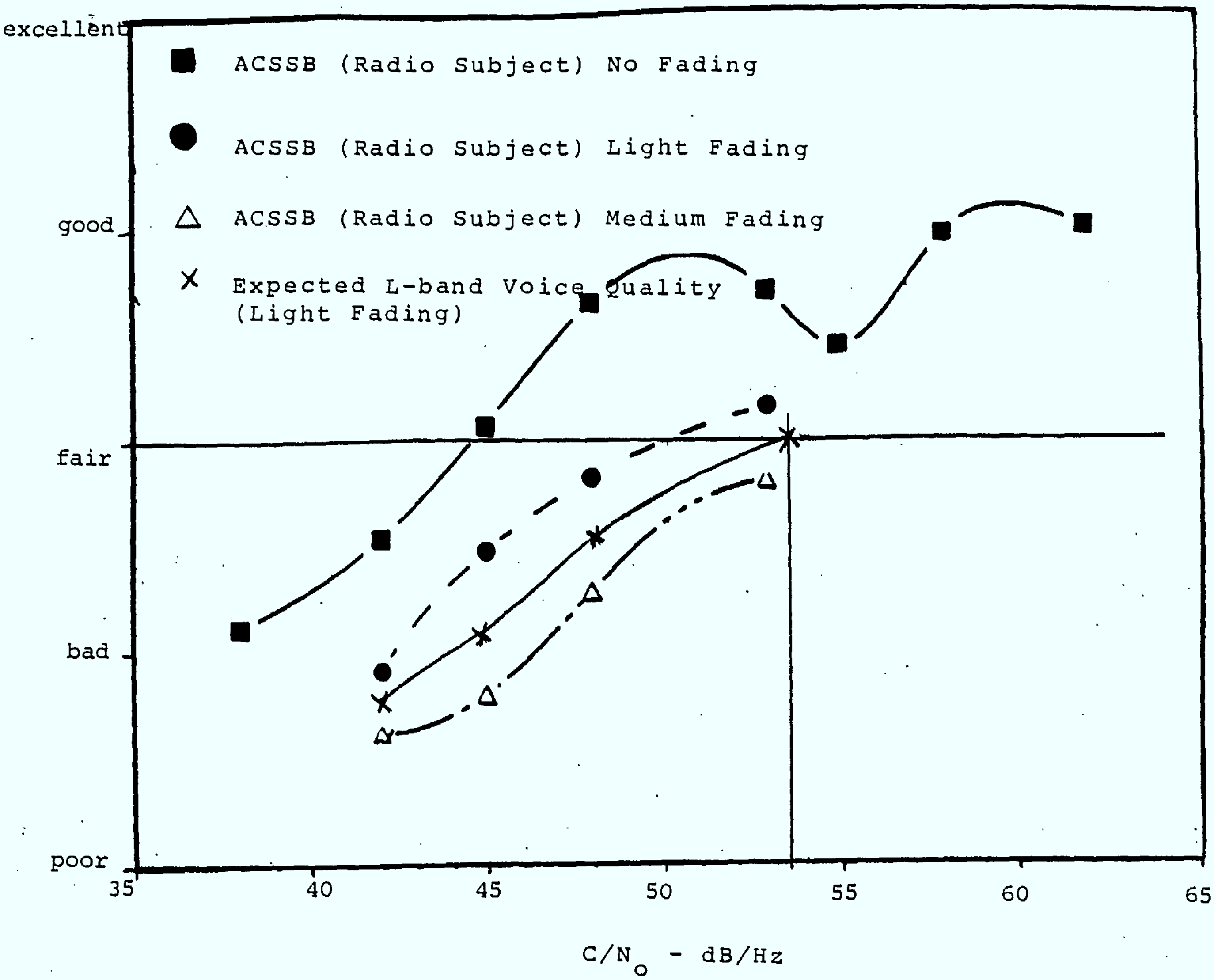


Figure 6.5: Subjective Quality of ACSSB With Fading

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CHAPTER VII: CONCLUSION AND DISCUSSIONS

7.1 Conclusion

The main objective of this task was to refine the mobile terminal specifications based on our current understanding of the relevant technologies and the previous studies conducted on this subject. That is, the specifications given here are by no means final and may change with any changes in the system design or advances in the related technologies. Therefore, in order to meet the target price for terminal hardware, while maintaining an overall performance consistent with the baseline system design, a dialogue should always be kept open with the equipment designers and hardware manufacturers until such time that the terminal design achieves a sufficient degree of maturity for implementation.

The BNR voice quality tests proved that in a mobile environment ACSSB performance would be acceptable to the general public. In addition, it demonstrated that further research and development is required to improve LPC vocoders performance.

A salient feature of ACSSB/TTIB is that the fading on the speech portion of the signal is correlated with fading on the pilot and that is necessary for good operation of the FFSR loop in the receiver. A companding ratio of 3:1 was favoured over a 2:1 ratio since an improvement in SNR of 5 dB could be achieved at the average speech level.

It has been reported that [3] the bandsplitting circuitry necessary for TTIB operation can be utilized to enhance the performance of the compandors used for subjective improvement of signal-to-noise ratio in speech channels. In general, the energy occurring in the upper and lower audio subbands is uncorrelated. Thus, independent noise suppression can be effective in both the upper and lower frequency subbands when the speech is present. A further advantage is obtained due to the fact that the noise bandwidth seen by each compandor is reduced by 3dB. Therefore, the saving in C/No may lead to a reduction in satellite EIRP requirements.

It appears that Adga experienced some difficulties in implementing the ACSSB terminal due mainly to the following reasons. First, perhaps for an effort to minimize the cost they selected low quality parts and their design called for many analog devices that needed frequent adjustment. For example, for accurate frequency references, they used RC oscillators which required adjustments each time the terminal was switched on. Second, their interference studies seem to be pessimistic and resulted in stringent constraints for RF filtering, the LNA, and IF filters. With relaxed interference suppression requirements (Appendix A), a designer would be able to meet the desired specifications with relative ease and, therefore, with an overall cost reduction of the terminals. With the reference frequency correction scheme, suggested and implemented by CRC, the local oscillator stability requirements are eased and probably can be achieved using inexpensive crystal oscillators.

The critical performance requirements for operation within MSAT were found to differ substantially from those of available terrestrial radio equipment. Therefore, a design based upon the existing FCC and DOC standards would be most probably not suitable for MSAT. Furthermore, it is perceived that modifying an existing FM or ACSSB terrestrial unit to operate in the MSAT system is not effective.

7.1.1 Full-Duplex Operation

The current terminal design is based on a half-duplex push-to-talk operation. Whenever the transmitter is in operation, the receiver is disabled and vice versa. However, in a full-duplex MTS operation, the transmitter and receiver could operate simultaneously. Since the mobile will most probably be using a common antenna for both transmit and receive operations, and since a perfect match across the band is not practically achievable between the antenna and transmission line, some of the radiated RF signal will be reflected into the receiver, and cause an overload of the low noise amplifier. Therefore, to protect the LNA from overload by the reflected high power transmitted signal, a very highly selective RF filter is required. This requirement might not be achieved easily at UHF or L-band if the overall terminal cost is to be kept in a reasonable range.

The need for a proper isolation between the transmitter and receiver is highlighted in the following example. The average radiated power of 11 dBW is reflected from the antenna as - 9 dBW, assuming 20 dB return loss for a practical antenna. The signal which is 45 MHz away from the wanted signal could be attenuated, for instance, by 40 dB (Tx-Rx isolation) at the output of the duplexer. Furthermore, the LNA protection filter attenuates the undesired signal by an additional 45 dB (typical of 2 - section RF filter). Hence, at the input of the LNA, two signals are present: the weak received wanted signal of -161.7 dBW and an interfering signal composed of the reflected signal at a level of -94 dBW and the leakage signal at -74 dBW. The presence of a high level interfering signal relative to the weak desired signal calls for high linearity of LNA with a low noise figure (See Appendix A).

7.1.2 LPC Performance

The BNR voice quality tests have shown that, even under ideal conditions, the LPC ratings were relatively low; also, the LPC vocoder was significantly affected by the background noise of a typical mobile which caused total loss of intelligibility.

The LPC vocoder designed at CRC is optimized for a male speaker for which the pitch period is greater than that of a female talker (see Appendix D for LPC operation). Therefore, the poor rating of the LPC in the subjective quality test is not only due to the subjects who were unfamiliar with the LPC's electronic accent, but also to the non optimality of the LPC algorithm. The results indicated a marked difference between the scores associated with male and female talkers. Seeking a solution to improve the performance of LPC must therefore, be considered. Such a solution could be the use of 4.8 kbps LPC.

The 4.8 kbps LPC vocoder appears to preserve better the characteristics of the original voice and reproduce it more faithfully. It can provide a method for generating natural speech as opposed to the traditional pitch pulse and white noise excitation used in the 2.4 kbps vocoder. The expected subjective improvement in SNR with the 4.8 kbps vocoder ranges from 2 to 5 dB depending on the type of talkers [4]. However, the 4.8 kbps LPC, would require additional bandwidth and power or more complex modulation techniques. That is, the 5 kHz channel spacing may not be sufficient to accommodate the coding schemes at a bit rate higher than 2.4 kbps. For example, DMSK is shown to have a degraded performance at higher bit rate than 2.4 kbps when limited in a 5 kHz channel bandwidth.

Recently, several new schemes such as trellis coding and multilevel PSK have been under study in the United States with some promising results.

7.1.3 MSAT Terminal Cost

Reducing the overall terminal cost is a major factor in the success of the MSAT system and a trade-off between terminal cost and equipment quality should be compromised. A modular design of the MSAT terminals is of special interest for ease of maintenance.

CRC, in their MSAT terminals design, demonstrated that narrow band mobile communications over a satellite link is technically feasible. The experience that CRC gained in this process would be of great value to the

Canadian industry in order to deliver cost effective terminals to the Canadian users at the time required, and would probably give them an opportunity to compete for part of the U.S. market.

The major contributors to the terminal cost, excluding the antenna, are the RF filters, the LNA, the balanced mixers, and the HPA. The various baseband processing operations could be implemented using signal processing chips like the TMS 32010 and TMS 32020. It is expected that a cost-effective terminal design could be achieved using ceramic loaded cavity RF filters as long as they are produced in large quantities [3].

7.1.4 Areas Requiring R&D

The current MSAT mobile terminal design, despite the development at CRC, is still far from being optimized. One has to bear in mind that a cost effective design meeting the specifications is of major concern for the success of MSAT. Therefore, further investigations should be undertaken in different areas in order to enhance the system's performance. Some of the areas which require further attention are:

a) Pilot Power Level

The level of the pilot required for TTIB is at the level of the average audio signal. Thus, the pilot would consume half of the satellite power inefficiently since the pilot does not carry any information. CRC has found that by reducing the pilot level the receiver performance is degraded. Therefore, under such conditions, it is required that the pilot be voice activated (40% activation). However, in the absence of the pilot the receiver would lose synchronization. Another alternative is to try to improve the FFSR loop operation in order to detect a low level pilot without any further complications.

b) Power Control For Mobile Terminals

The MSAT baseline design assumes that a mobile is operating at the edge of the service area. Based on this assumption, the required transmitted EIRP from a mobile was determined to be 11 dBW. However, for mobiles operating close to the centre of the beam, the satellite G/T could increase by as much as 3dB. If the transmitted power from those mobiles operating near the centre of the beam were not adjusted properly in order to compensate for the higher satellite G/T, their signals would benefit from a higher satellite power, at the expense of those mobiles located away from the beam centre, as well as causing extra intermodulation interference. Thus, a cost effective scheme should be developed to control the transmitted power from a mobile terminal. The controller could also be used when the mobile is suffering from multipath and shadowing losses.

c) Data Transmission Application

Recent studies demonstrated the need to establish data transmission services within the MSAT network. The type of services could range from a low bit rate (about 300 bps) to a higher bit rate (2.4 kbps), each requiring different system design considerations like the access techniques, the message length, the coding method, etc. Also, the system throughput, and the delay associated with a successful transmission would constitute a major factor in the overall cost of the system. This issue should be further investigated and, as a result, the data terminals hardware and specifications should be refined.

d) Subjective Evaluation of Speech

A new approach to estimate the subjective quality of speech has been proposed recently by ADGA. In this method, the subjective quality is found to be correlated with the average signal-to-noise ratio in the channel. This is accomplished by subdividing the audio band into several subbands, measuring the energy in each subband, then forming a weighted average which gives an estimate of the subjective quality. Therefore, the method does not deal only with the energy content of each subband but rather with the sensitivity of the human ear to the variation of power in each subband. A close investigation of this method is required along with the degree of correlation with the traditional methods.

e) Adjacent Channel Interference

An important factor of a receiver performance is its adjacent channel selectivity. It is defined as a measure of the capability of a receiver to receive a wanted signal without exceeding a given degradation due to the presence of an unwanted modulated signal. The undesired signal differs in frequency from the wanted signal by the spacing between adjacent channels for which the equipment is designed. Although this measurement could be done in a laboratory, the subjective evaluation of the adjacent channel selectivity is desirable. No such test has been yet conducted for an ACSSB receiver. The CRC experience has shown that the companding which is a non linear process, significantly reduces the intelligibility of the adjacent channel interferer. However, the case of interest is especially when the wanted channel is faded and the adjacent channel is not. This situation is likely to happen in an MSAT environment, especially when the adjacent channels are assigned to users located in different parts of the coverage area.

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APPENDIX A

ACSSB Terminal Requirements

A.1 Terminal Critical Design Problems

The expected specifications of ACSSB terminals given in this report were based on a cost effective design with the appropriate characteristics.

Note that the terminal cost is mainly due to the RF parts. Therefore, an optimization of the RF components is required. For example, the design of Class A HPAs should be optimized to reduce the harmonic distortions and spurious emissions which fall into the MSAT channels as well as into the terrestrial bands.

The following addresses briefly the requirements of different parts of an ACSSB MSAT terminal.

A.1.1 Receiver RF Filtering Requirements

The received signal level from the satellite is:

$$C = EIRP - L_p + G$$

where EIRP is the UHF satellite EIRP per carrier, L_p is the free space path loss and G is the mobile antenna gain. For EIRP = 26.5 dBW, L_p = -183.2 dB and G = 8 dBic, the carrier power will be -148.7 dBW. Assuming 13 dB of fade margin for 99% of signal availability, the faded signal level would be -161.7 dBW.

The aim of RF filtering is to suppress the interfering signals of adjacent bands to MSAT and to protect the LNA. Table A.1 shows the strength of interfering signals, emitted from terrestrial stations, appearing at an antenna input with 0 dBi gain, where the specified ranges are the same as used in the ADGA study [1].

It appears that due to the small distances assumed in [1] from the sources of interference, most of the expected level of interfering signals are unrealistic. Note that the MSAT objective is to provide mobile communication services to rural areas which are not served by any other communications means. Table A.1 shows that the interfering signal from a trunk base station, which operates adjacent to the MSAT receive band, is at a level of -35 dBm when the MSAT terminal is only away by 0.5 km from the base station. In reality, such close proximity will rarely occur and by changing the range to 10 km, the level of interfering signal will be reduced to -61 dBm.

The above discussion suggests that the constraints put on the RF Filtering requirements by ADGA are unrealistic and should be relaxed. The required RF filtering for the worst case situation, in order to provide a signal-to-interference ratio of 10 dB, could be accomplished by using a 2 cavity RF filter [1].

Figure A.1 shows the characteristics of a 1-section and a 2-section filter. Based on ADGA's study, 28 dB of attenuation is required at 21 MHz from the centre frequency, while at 41 MHz away, the RF signal will be attenuated by 40 dB. However, with modified ranges for

Source	Freq. Band [MHz]	Freq. Sep. [MHz]	EIRP [dBW]	Range [meters]	Received Power in dBm
Class B UHF TV1	600-806	-60	50	1000	-9
Class C UHF TV	600-806	-60	60	1000	+1
Trunk Mobiles	806-821	-45	14	20	-11
MSAT Mobiles	821-825	-41	11	5	-6
Cellular Mobiles	825-845	-21	6	20	-17
Trunk Base	845-866	-0	24	500	-35
Cellular Base	870-890	+0	21	1000	-41

Table A.1: Interfering signals with specified ranges.

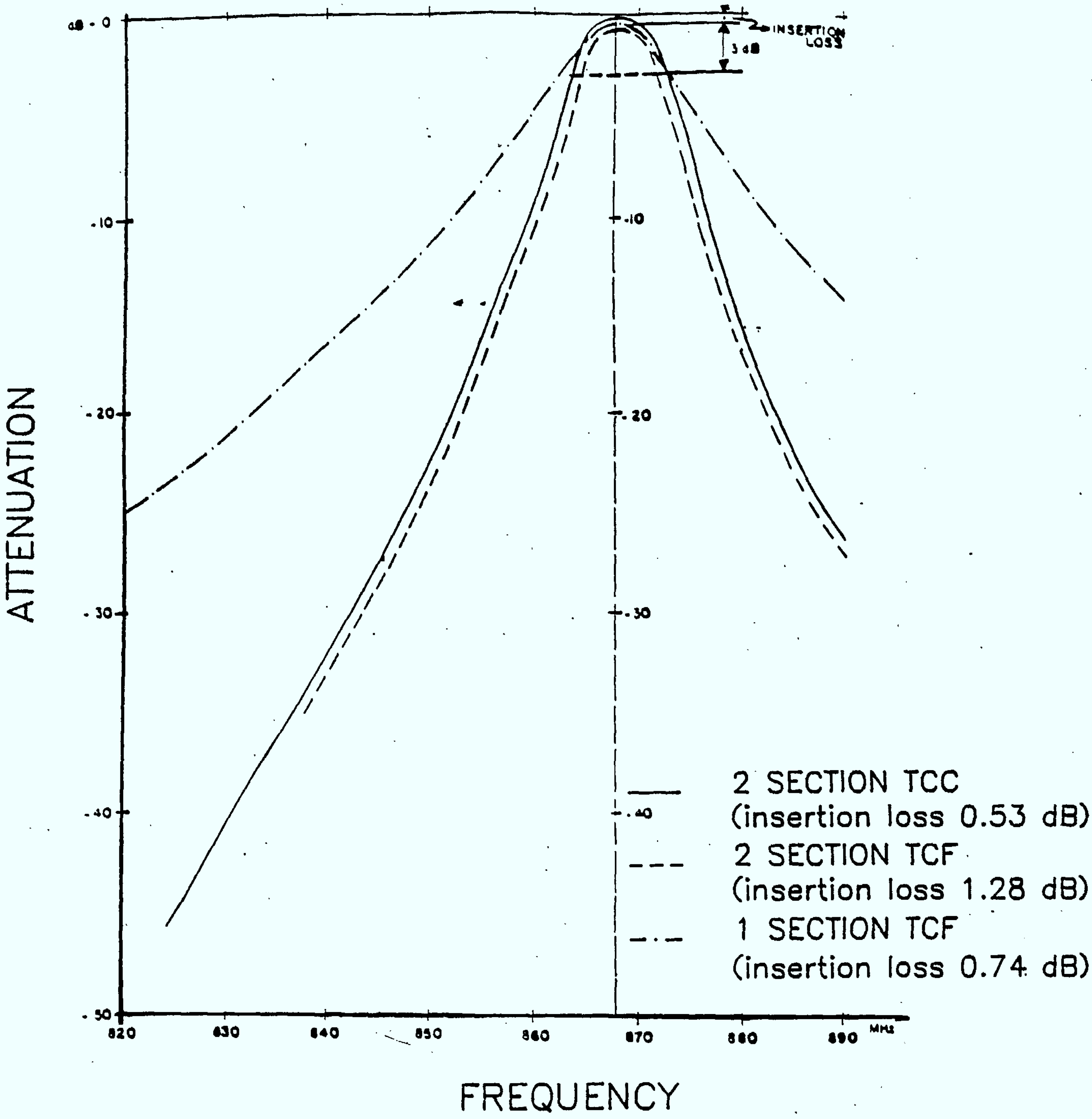

Figure A.1 : Frequency Response of RF filters.

Source	Freq. Sep. [MHz]	Range [meters]	Received Power [dBm]	Required Attenuation [dB]
Class C UHF TV	-60	10,000	-19	46
Trunk Mobiles	-45	80	-23	42
MSAT Mobiles	-41	50	-26	39
Cellular Mobiles	-21	80	-29	36
Trunk Base	-0	5,000	-55	-
Cellular Base	+0	10,000	-61	-

Table A.2: Interfering signals with modified ranges.

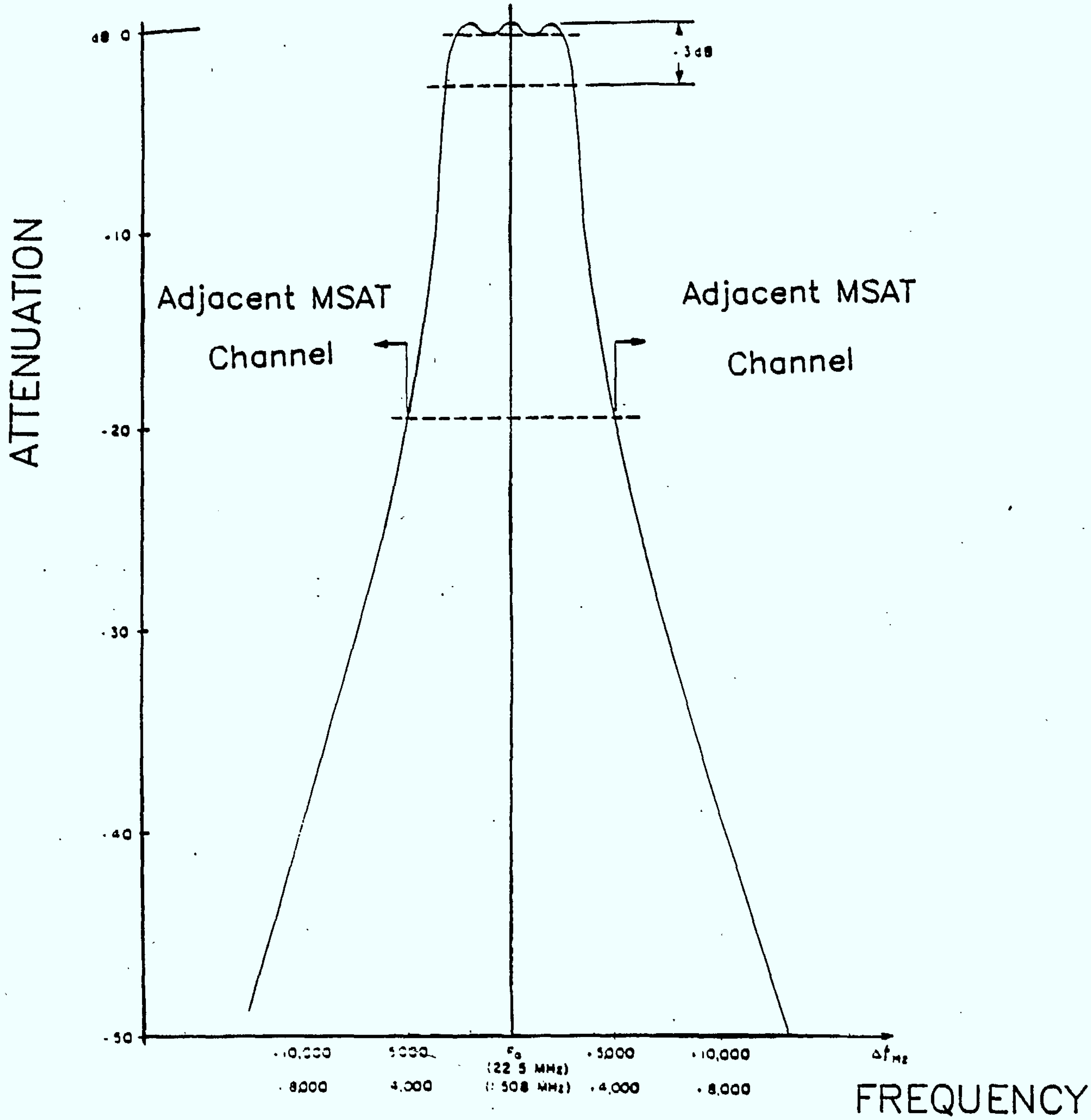

Figure A.2 : Frequency Response of a 3-pole Butterworth filter.

interfering signals, the 2-section RF filter could be replaced by a one-section filter. The interfering signals with the modified ranges appear in Table A.2.

The price of a one cavity RF filter with 0.7 dB insertion loss is quoted to be about U.S. $100 and rises up to $900 with a 2-cavity filter and 0.7 dB insertion loss [1]. Note that one may not be able to achieve a low receiver noise figure unless front end filtering is distributed around the LNA. Therefore, it has been suggested that two RF filters be used; the pre-LNA and post-LNA filters. The noise figure of the receiver is mostly dependent on the receiver front end components, i.e. the pre-LNA filter, the LNA and the downconverter. RF filters should have, as a minimum, a 3 dB bandwidth of 4 MHz, and a low insertion loss.

A.1.2 Low Noise Amplifier (LNA)

The main function of the LNA is to provide amplification of low-level signals received from the satellite in the presence of high-level interfering signals from nearby terrestrial sources. The minimum frequency range of the LNA is 4 MHz (866 to 870 MHz).

The front end RF filtering will not be able to reduce sufficiently the interference from the bands adjacent to the MSAT channels. Hence, the IF filters should reject the unwanted signals.

Figure A.2 shows the characteristics of a 3-pole Butterworth IF filter. As seen in the figure, 20 dB of attenuation is accomplished at 5 kHz away from the first MSAT channel. At 15 kHz away, the interfering signal will be attenuated by 50 dB. The interference from a trunk base station adjacent to the MSAT band is 100 dB higher than the desired signal when the MSAT receiver is at a 500 meters range. However, by using the relaxed interference requirement as given in Table A.2, 75 dB of attenuation will be required, hence a 4-pole Butterworth filter, instead of a 6-pole Butterworth filter suggested by ADGA, is expected to be adequate.

A.2 Transmitter Characteristics

A.2.1 Transmitter Average Radiated Power

The EIRP per carrier given in the Telesat link budget is 11.0 dBW [7]. For an 8 dBic antenna gain, the emitted power would be 3.0 dBW. For 4 dBic gain it will become 7 dBW which could increase the cost of the HPA significantly. Note that the terminals with 4 dBic gain are expected to operate in areas with insignificant shadowing loss and, therefore, they may not need as much transmitted power.

A.2.2 Transmitter Third Order Intercept

The third order intercept point of an amplifier is given by [8]:

$$IP(3) = (2P_1 + P_2 - IM(3))/2$$

where P_1 and P_2 are the power levels of two signals at frequencies f_1 and f_2, respectively, with IM(3) as their third order intermodulation product. In a single channel, the IM products of the audio signal and the pilot are important. Here, we are only concerned with the third order IM. Assuming the audio-signal and the pilot reference are at equal power level, then, $P_1 = P_2 = 3$ dBW - 3 dB = 0 dBW = 30 dBm (this is for an antenna gain of 8 dBic and EIRP of 11 dBW).

Telesat's baseline design assumes that the total of IM and energy spreading is at least 25 dB below the carrier. Here, we assume that the required IM product is 30 dB. The peak average power is what we should really consider in calculating the third-order intercept. For the conventional 2 tones at 1.0 kHz and 1.7 kHz respectively , the peak average power is about 37 dBm, therefore, the third order intercept point is at:

$$IP(3) = (74 + 37 - (37 - 30))/2 = 52.0 \text{ dBm}$$

Note that a better IM product could be achieved with an expensive high power amplifier. For example, the terminal designed by CRC has an HPA with the third IM response 40 dB below the average power.

A.2.3 Transmitter Pilot Power Level

The pilot reference level is at the average audio signal level. Although it has been assumed earlier that the pilot level would at least be 10 dB below the average, the CRC experience has indicated that it should be at about the average audio level in order to recover the pilot at the receiver. Therefore, the pilot should be voice activated to save satellite power.

A.2.4 Transmitter SSB Suppressed Carrier

The carrier in the SSB signal should be suppressed sufficiently to save satellite power. Filtering is used usually to suppress it. A typical value of carrier suppression would be 30 dB below the average power.

An alternative scheme for generating a single-sideband signal is based on a phase discrimination method [8]. Here, two balanced modulators are employed. This scheme generally enjoys less popularity than does the filtering (frequency discrimination) method. The reason is that the present phasing method requires several stringent constraints, if the carrier and one of the sidebands are to be suppressed adequately.

Note that the CRC design is based on the former method.

A.2.5 Transmitter Frequency Response

The transparent tone-in-band (TTIB) signal occupies 3.9 kHz of bandwidth. The RF filter and the HPA in the transmitter chain should have a minimum bandwidth of operation of 4 MHz. The 3 dB bandwidth of the transmitter, assuming an audio input signal at baseband from 300 Hz to 3000 Hz, will be about 5.0 kHz. Figure A.3 shows the transmitter frequency response of the CRC terminal.

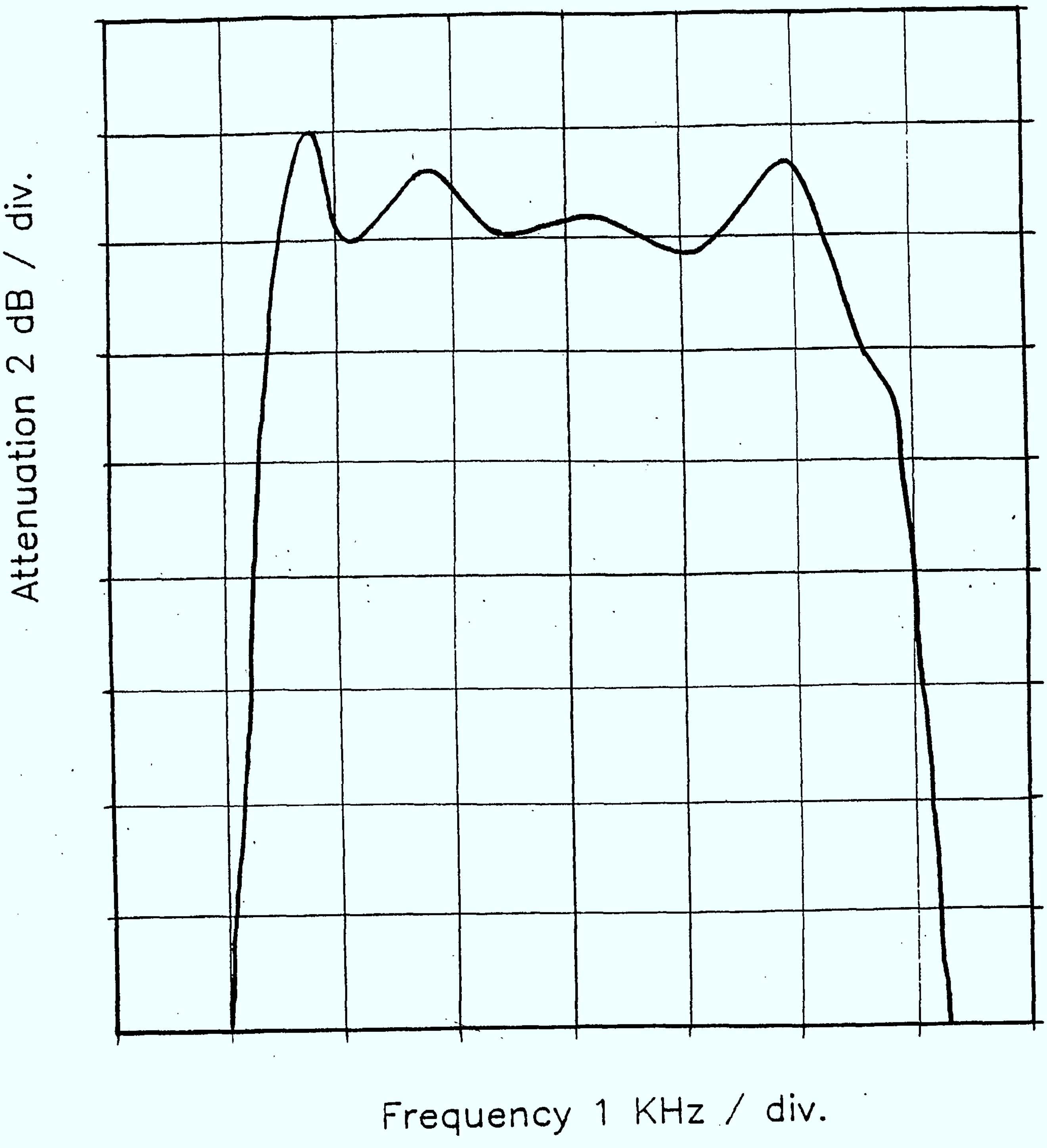

Figure A.3 : Transmitter RF frequency response.

A.2.6 Transmitter Adjacent Channel Spillover

IF filters are assumed to limit power spillover on the adjacent channels. The amount of energy spread in the Telesat link budget is about 28 dB below average power. That is, the power measured in 5 kHz bandwidth, contributed from an adjacent channel will be 28 dB below average power. The proposed value here is 30 dB in order to have some margin.

A.3 Receiver Characteristics

A.3.1 Receiver Noise Figure

The receiver noise figure is defined as the ratio of the SNR at the input to the SNR at the output. In the receiver chain, the main contributors to the noise figure are the front end parts such as the RF filter, the LNA, and the downconverter.

The current Telesat baseline system design is based on a receiver with a 2 dB noise figure. The present CRC in-house design also has a 2.0 dB noise figure. Note that ADGA could only achieve a noise figure of 2.1 dB due to the fact that they put very stringent requirements on the IF and RF filterings, as it was discussed in Section A.1.1.

A.3.2 Receiver Sensitivity

The receiver sensitivity is a measure of the minimum signal level which can be detected by the receiver, and it is related to the total noise floor in the system. Assuming a noise bandwidth of 3.9 kHz and a total system noise temperature equal to 510K, results in a noise power of -135.6 dBm. Therefore, it is desirable that the sensitivity of the receiver be in the vicinity of the noise floor.

A.3.3 Receiver Audio Frequency Response

At the baseband output of the receiver, the audio signal from 300 Hz to 3 kHz should be regenerated. The 3 dB bandwidth, as shown in Figure A.4, will therefore be from 300 Hz to 3000 Hz. The response above 3000 Hz will roll-off monotonically and will be no higher than -25 dB at 4 kHz.

A.3.4 Receiver Third Order Intercept

According to Table A.1, the signal in the band immediately adjacent to MSAT at levels up to -35 dBm would not be reduced by the front end filtering. Since the faded signal level is about -139.7 dBm at the antenna input, therefore, the third order intercept at the output of the LNA (IP(3)) with 20 dB gain is [1]:

$$\text{Input level (I)} + \text{LNA Gain} + [\frac{\text{Interfering Signal Level}}{\text{faded signal level}}$$

$$+ \text{Desired Signal to Interference Ratio}]/2$$

$$IP(3) = -35 + 20 + (-35 + 139.7 + 10)/2 = 42.35 \text{ dBm}$$

This LNA requirement is the result of a worst case interference assumptions (as stated by ADGA). In the case where the signal level in the band adjacent to MSAT is relaxed to -50 dBm, the third order intercept at the output of the LNA would be about 20 dBm.

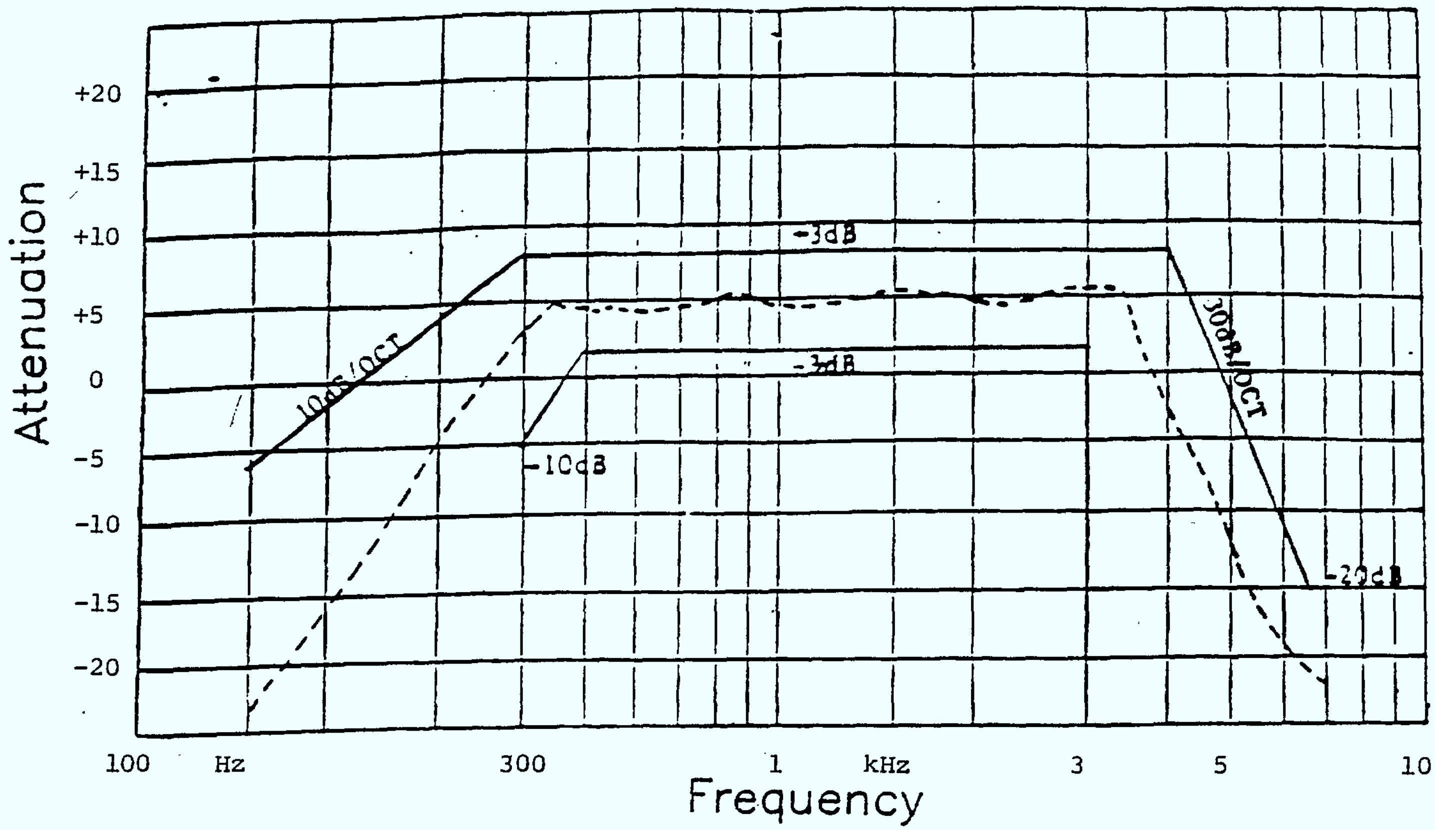

Figure A.4 : Receiver Audio frequency response

A.4 Frequency Synthesizer Characteristics

A.4.1 Synthesizer Switching Speed

The synthesizer should be able to switch at 5 kHz steps over the 4 MHz transmit and receive bands. The synthesizer switching speed and the time the oscillators take to lock on the required corrected frequency must be less than the inherent delay between the instant an acknowledgement is sent to DAMA and the time DAMA receives it. Assuming a one-way delay of 270 msec and a frequency lock time of 100 msec, the synthesizer switching speed should be about 150 msec [2].

A.4.2 Oscillators Frequency Stability

The mobile, being equipped with an AFC loop, will be able to lock on to the carrier in the range of ± 440 Hz [2]. Therefore, the frequency stability required after the first frequency acquisition is:

$$\pm 440/(870 \times 10^{6}) = \pm 0.505 \text{ ppm}$$

However, the aging effect imposes a more stringent constraint on the overall terminal frequency stability, hence ± 0.1 ppm is proposed. Note that CRC with their frequency distribution plan have been able to accomplish less than ± 0.1 ppm. However, ADGA was able to obtain only ± 0.3 ppm. Several manufacturers were contacted during the course of this study. The responses received indicate that with low cost oscillators, stability is very difficult to achieve at the present time. Thus, if no progress in L.O. design is accomplished in the very near future, the required terminal stability should be somewhat relaxed. In addition, the extent of the aging effect on the stability of the L.O.s in a mobile environment should be investigated.

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APPENDIX B

Record of Conversations with Mobile Terminal Manufacturers

In the course of the study, several North American mobile terminal manufacturers were contacted in order to obtain information on their current and future products, as well as to explore their plans in the design and manufacturing of MSAT mobile terminals. The organizations contacted are:

- Glenayre, Vancouver, B.C.
- Novatel, Alberta
- Stephens Engineering Associates, Seattle, Washington
- E.F. Johnson, Wasica, Minnesota
- Aeroton (Sideband Technology), Raleigh, North Carolina
- Contemporary Communications Company, New Rochelle, N.Y.

The following gives a summary of this survey.

B.1 Glenayre Ph.# 604-293-1611
Contact: David Loop

Currently, Glenayre's main products are the radio control heads for mobile terminals. The RF/IF parts of the radio are purchased from Motorola and Canadian General Electric (CGE). The company's principal unit is the 2020 mobile radio which operates at VHF and at 800 MHz UHF. The unit uses FM modulation with 25 kHz channel spacing. The HPA power output is 25 watts RMS. The radio operates in accordance with the Radio Standard Specification (RSS) 119.

Their transceiver units are off-the-shelf units. Glenayre is planning to design and manufacture their own RF/IF segments of the terminal. The design of control heads for the terminals will still be pursued, whereas UHF and VHF transceivers will be their main products in the near future.

Glenayre is definitely considering the introduction of the narrow band technology into the mobile communications market. Existing ACSSB services are mobile radio services (MRS half duplex). No product has so far been introduced in the full duplex telephone service. Glenayre has recently been negotiating a contract with the Canadian Department of Communications to design and build a prototype ACSSB terminal. They believe that some of the current CRC terminal specifications are difficult to meet; e.g. the frequency stability of $\pm$0.1 ppm could be realized only with relatively expensive local oscillators. A stability of $\pm$ 0.5 ppm is the most likely value to be obtained in the MSAT ACSSB terminal with a reasonable cost. The work was expected to start at the end of April, 1986. Glenayre is awaiting the Government's decision on financial support in order to pursue its MSAT terminal studies.

In 1983, under a CRC contract, Glenayre tried to transform an FM mobile terminal to meet the MSAT requirements. However, they did not succeed mainly due to the RF section requirements. Their final recommendation was a separate design for MSAT terminals.

B.2 <u>Novatel</u> PH# 403-298-0465
Contact: Clement Li

Novatel is presently marketing terrestrial mobile communication equipment at UHF frequencies. Their radio transmit and receive frequencies are 825 MHz - 845 MHz and 870 MHz - 890 MHz, respectively. They also have a few products operating in the lower part of the UHF band, like the Aurora 400 system. Frequency modulation is used for voice communications while Frequency Shift Keying (FSK) is employed for data transmission. The RF output power is 3 Watts continuous.

Novatel will continue the production of UHF mobile terminals using frequency modulation. They will also be enhancing their current product lines and adding many control features to the terminals.

The narrow band technology has not been considered by Novatel yet. They believe that FM out-performs ACSSB in terms of voice quality and design simplicity. The FM requirements are less stringent than those of ACSSB which, according to Novatel, can hardly meet the required intermodulation response in the cellular systems.

Novatel believes that they can design and manufacture terminals to satisfy MSAT specifications if the Government supports them in this program.

The requirements for MSAT ACSSB terminals and those of the terrestrial systems are totally different. They feel that FM and ACSSB are not compatible at all. First, the terrestrial mobiles are exposed to Rayleigh fading while the MSAT environment is characterized by a Rician fading and shadowing. To experience 35 to 40 dB of fade depths in terrestrial mobile is not uncommon, while the fade depth encountered by MSAT mobiles does not usually go beyond 20 dB. That is, the fade correction circuitry in cellular should have a larger dynamic range.

MSAT requires a very sensitive receiver due to the weak signal received from the satellite. The frequency stability used currently by cellular terminals is not likely to be acceptable for MSAT. MSAT terminals need a 0.1 ppm stability after acquisition of the reference pilot.

B.3 Stephens Engineering Associates

PH#: 206-771-2182
Contact: Norm Shirley

The main product of this company is the SEA 175 ACSSB radio which operates at VHF frequencies in the band 150 to 158 MHz. It is equipped with 16 half duplex channels with 5 kHz spacing. The transmitter power rating is 25 Watts PEP. Using the Tone Above Band (TAB) technique, the pilot is placed above the audio frequencies at 3.1 kHz. The pilot level is 10 dB below PEP when no modulation is applied and 18 dB below PEP with full modulation.

Many design considerations have to be re-evaluated in order to convert the SEA 175 radios to UHF MSAT radios. The main problem they see is maintaining the MSAT terminal frequency stability requirement. Their current oscillator's stability is 2 ppm . Also, they see a change in the baseband processing where TAB must be replaced by TTIB, to correlate the fadings on the pilot and the audio signal. SEA also said that a terrestrial mobile would experience fading depths as deep as 35 to 40 dB while an MSAT UHF mobile will not be exposed to higher than 20 dB fade depth. This reduction in fade depth is in the advantage of MSAT design since the AGC in the receiver can operate in a narrower dynamic range.

SEA believes that pilot pumping is inefficient over a satellite link due to reduced dynamic range of the signal level. That is, the pilot may fall below the noise floor causing distortion in the receiver. Also, no gain in IM reduction could be extracted when the pilot is 10 dB below the peak power.

Since the receiver requirements at UHF are different from those at VHF, the front end part of SEA receiver including LNA and RF filters should definitely be redesigned.

B.4 E.F. Johnson

Phone number: (507) 835-6378
Contact: Robert Brandell

E.F. Johnson manufactures radios operating in frequency bands of 30-50 MHz, 150-170 MHz, and at UHF frequencies in the band 450-512 MHz and at 900 MHz (airfone). They have recently built and tested a VHF prototype unit operating at 150 MHz with TTIB processing and incorporating FFSR in the receiver for frequency and fade correction. This unit would tolerate ±1.25 ppm as the local oscillator frequency error. The pilot is centred in the variable notch width at a level 8 to 10 dB below peak envelope power of 25 Watts. E.F. Johnson supports the idea that TTIB is preferred over TAB at UHF, although many changes are required to make their own unit operate within the MSAT system.

The main requirements at UHF are:

- Design of a very sensitive receiver for satellite signal. That is, the receiver should have a low noise figure.

- Sharp RF and IF filters to suppress adjacent band interference.

- TTIB with FFSR processing.

- Linear power amplification to minimize IM distortions.

- A frequency synthesizer with high switching speed and accurate frequency lock range.

B.5 Aerotron, Inc

Phone Number: (919) 872-4400

Contact: Harold Davis/Doug Collette

This company produces a series of ACSSB radios operating at VHF, 136-174 MHz band. Their product line includes trunk mount mobile transceivers, desk-top base stations, dash mount mobile transceivers and base/repeater stations. Their receivers have 5 kHz spacing, 25 Watts PEP, and $\pm$ 0.2 ppm frequency stability with AFC. The receiver sensitivity is -117 dBm. They feel that designing a separate unit for MSAT will be more beneficial, since an optimal frequency distribution is necessary when operating at 800 MHz.

B.6 Summary

The survey of the manufacturers contacted indicates that designing a new unit for MSAT will be more cost effective than transforming either the current VHF ACSSB terminals to operate in MSAT system. The 900 MHz airfone terminal technology could be expanded to MSAT although it is based on 6 kHz channel spacing and it is compatible single side band (CSSB) not ACSSB. The following aspects of the current ACSSB terminals should be redesigned for MSAT application:

- Receiver sensitivity
- IF and RF filtering
- Baseband processing
- Receiver noise figure
- Frequency synthesizer switching speed

CRC has proven the feasibility of designing 5 kHz ACSSB radios at UHF which can combat MSAT fading conditions. However, the cost of their radio is relatively high. Therefore, the cost optimization is of a great importance in future studies. A list of specifications for VHF and UHF ACSSB radios appears in appendix E.

APPENDIX C

THE MSAT MOBILE CONTROL UNIT

The guidelines for the design of an MSAT terminal control unit must be determined based on considerations of driving behaviour, customer preference, automobile environment, and calling procedure. In this appendix, we explain these guidelines and illustrate their application in the design of a control unit.

The control unit should give users access to MSAT communications service in a moving vehicle. However, formulating control unit design guidelines involves meeting a challenge not encountered in the PSTN experience. The vehicular nature of this service requires that the control units have negligible effect on driving behaviour and conform to the automobile's environmental constraints.

C.1 Driving Behaviour

Mobile telephone users are commonly engaged in additional activities while driving. Such activities include talking to passengers and adjusting the car radio, etc. Although these activities cause some distraction, it seems that the driver gives priority to driving demands and tries to minimize the effects of secondary activities on driving behaviour. The use of an MSAT terminal is viewed as a secondary activity similar to those described above. Therefore, to minimize the risks involved in operating an MSAT terminal, the control unit should include several options related to placing a call, like automatic call placement and the hand-free operation. Also for ease of use, MSAT users would most probably prefer controls that are mounted near the top of the instrument panel and to the right of the steering wheel, as in the terrestrial radio and telephone services. Note that the protection of the unit against theft and vandalism would also be of major concern. Thus, the location of the unit should be inconspicuous.

C.2 Automobile Environment

While users would prefer that the control unit be built into the instrument panel, this is not likely to be the case since tightly spaced controls, indicators and optional devices such as radio and clocks occupy most of the usable space within reach of the driver. A flexible, adaptive approach to mounting is required to position the control unit in locations acceptable to the driver.

Control units should be designed to give service to ambient temperatures as high as 85°C. Traditional plastic housings and handsets do not perform well at this temperature. Thus, high temperature material should be used for the control unit.

Measurements made for the AMPS show a high ambient noise level in the automobile environment. At 90 km/hr on a smooth road, ambient noise levels of about 70 dBA have been observed. This is roughly similar to the noise found in a busy office. With the automobile parked in a quiet location, the noise level drops to 40 dBA, the level observed in a quiet room. This wide variation in background noise level suggests that user-adjustable level controls are needed for the received audio signal.

C.3 Calling Procedures

To place a call, the user dials the ID number of the called party into a storage register within the mobile unit where it is automatically held until the user presses a send function key. When this key is pressed, the mobile unit initiates a call attempt by sending a digital message, which includes the ID numbers for both the calling and the called party, to the DAMA controller on the request channel. The stored number must be displayed in order to allow the user to check it and, in the case of dialing error, to cancel it. The number could be displayed by means of an LCD shielded display. If the transmission to DAMA is successful, the latter will respond with the assigned transmit and receive frequencies to establish the call. To terminate the conversation, the user goes on-hook.

C.4 Control Unit Functional Characteristics

The functional characteristics of the control unit are summarized in the following:

- Touch Pad Dialing: A standard full-size illuminated touch pad insures maximum ease and speed in dialing.

- Called Number Display: The called number is displayed on a large 11-digit display that is easy to read even under bright sunlight conditions.

- On-Hook Dialing: For user safety, the called telephone number is keyed into memory and stored prior to transmission. The number is retained, after transmission for last number recall.

- Automatic Call Placement: After entering the number to be called, pressing the send key will initiate a call. The transmitter "on" indicator is illuminated. Only when the called party answers, the voice channels are assigned to both parties. If the called party is busy, the busy indicator is illuminated.

- Stored Incoming Calls: When the user is logged-on and is away from the terminal, and an incoming call occurs, the terminal logic unit should store the caller phone number and the time that the call occurred in a buffer and activate a flashing light. When the user returns to his car, he will be able to retrieve the stored information by depressing the necessary function key.

APPENDIX D

THE LINEAR PREDICTIVE CODING OF SPEECH

D.1 Introduction

Predictive coding is an efficient method of converting signals into digital forms. The coding efficiency is achieved by removing the redundant structure from the signal before digitization. The predictor forms the estimate for the current sample of the input signals based on the past reconstructed values of the signal at the receiver [1] - [3].

Due to the limited resources in MSAT, the 2.4 kbps pitch-excited LPC appears attractive for voice communications. However, the implementation of the LPC algorithm which needs to be optimized for all types of speakers, is costly despite the fact that high speed digital signal processors are commercially available [4], [5].

In general, vocoders can achieve very low bit rate transmission of speech. The most significant accomplishment in the area of vocoded speech has resulted from the use of a concatenated tube model of the vocal cavity. This model possesses properties which are ideally suited to a digital analysis and synthesis of speech messages. It will be shown that the vocal cavity can be represented by a concatenated arrangement of tubes of different areas but of identical lengths. The tube model parameters are related directly to the coefficients of a linear prediction which models speech in terms of a weighted linear combination of past speech samples. The coefficients of the predictor are obtained by minimizing the sum of the squared differences between the actual speech samples and the predicted value over a finite interval. A typical linear predictor will utilize ten coefficients to control a tube model with ten tube sections.

D.2 Concatenated Tube Model of Vocal Tract

The concatenated tube model of the vocal cavity is shown in Figure D.1. The cross section of the vocal cavity is depicted in Figure D.1a, while the tube model of the vocal tract is shown in Figure D.1b. Note that the tube model consists of concatenated tube sections all having the same length ΔX but with different areas to match corresponding sections of the vocal tract [6].

Let C be the propagation velocity = 35000 cm/s, and T be the time delay per section = $\Delta X/C$. From the Nyquist sampling theorem, the Nyquist sampling period must be equal to or less than the reciprocal of twice the maximum frequency. Hence, the propagation time for each tube section of the model must be equal to or less then the reciprocal of four times the maximum frequency period. For f_{max} = 4 kHz, the value of T must be equal to or less than 62.5 microseconds, and the corresponding value of ΔX must be equal to or less than 2.1875 centimeters; thus, 8 tube sections are needed to model the length of the vocal tract which is approximately, 17.5 cm. If the maximum frequency were 5 kHz, ten sections would be needed.

The system's transmission response can be expressed in terms of reflection coefficients at each interface between tube sections. The relationship for the reflection coefficient r_k at the interface between tube section k

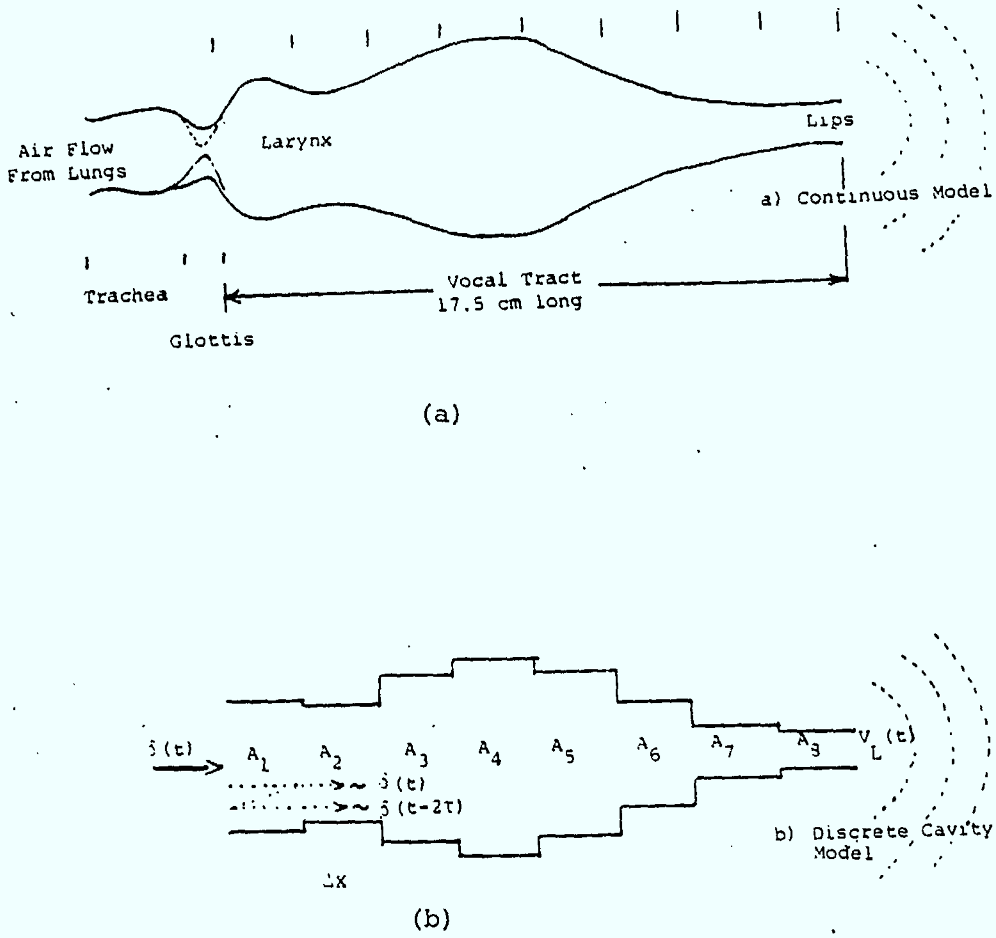

Figure D.1- The cross section of the vocal cavity (a) and the tube model of the vocal tract (b) and tube section k + 1 is given by the ratio [6]:

$$r_K = \frac{A_{k+1} - A_k}{A_{k+1} + A_k} ,$$

Where A_i is the area of section i.

Thus, if the tube sections on both sides of the interface are of equal area, the reflection coefficient is zero. The impulse response of the tube model is obtained by applying a Dirac impulse to its input and determining the amplitude of the impulses emitted from the output as a result of internal reflections. The impulse response is:

$$H(z) = \frac{Q(z)}{D(z)}$$

Where Q(z) is not frequency dependent and represents only tube transmission loss.

$$Q(z) = 0.5\ (1 + r_G)\ \sum_{k=1}^{N} (1 + r_k)\ z^{-N/2}$$

$$D(z) = 1 - \sum_{k=1}^{N} \alpha_k\ z^{-k}$$

Where, r_G is the reflection coefficient at the Glottis (see Figure D.1), N is the number of sections and, α_k are coefficients determined by the reflection coefficients. α_k are obtained by using the recursion formula:

$$Do(z) = 1$$

$$D_k\ (z) = D_{k-1}(z) + r_k\ z^{-k}\ D_{k-1}\ (z^{-1}),\ k= 1,2,\ ...,\ N$$

$$D(z) = D_N(z)$$

D.3 Relationship between Linear Predictor and Tube Model

A general diagram of the speech production model is shown in Figure D.2. Either voiced or unvoiced excitation produces samples of u(n) which are acted on by the transmission response of a vocal cavity model to produce signal output samples S(n). The transmission response of this process is:

$$H(Z) = \frac{S(Z)}{U(Z)} = \frac{G}{1 - \sum_{k=1}^{P} \alpha_k z^{-k}}$$

Where G is a gain factor which replaces the numerator term and is made proportional to the signal intensity. The denominator is the important term since it represents the spectral information content of the speech. An expression can be also developed for the error of the linear predictor in terms of its signal input.

A linear predictor with coefficients a_k produces an estimate of speech samples [6]:

$$s(n) = \sum_{k=1}^{P} a_k \, s(n-k)$$

The prediction error e(n) is given by:

$$e(n) = S(n) - P(n) = S(n) - \sum_{k=1}^{P} a_k \, s(n-k)$$

The error transfer function A(z) is:

$$A(z) = 1 - \sum_{k=1}^{P} a_k z^{-k}$$

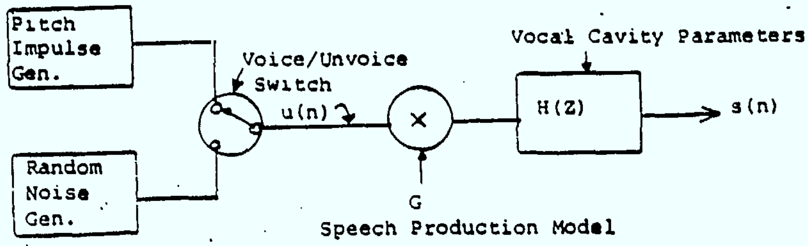

Figure D.2- The general diagram of the speech production model.

Therefore:

$$H(Z) \cdot A(Z) = G$$

This important relationship is the basis of the linear predictive vocoder concept.

D.4 <u>Structure of the Linear Predictive Vocoder</u>

Figure D.3 shows the block diagram of a complete linear predictive vocoder. At the analyzer, a linear predictor is used to determine the coefficients a_k and to convert these to the reflection coefficients r_k. Eight to ten such coefficients are used. In addition, the analyzer also contains a pitch detector for extracting the fundamental frequency of pitch, a voiced/unvoiced detector for selecting pitch or noise excitation and a signal intensity detector for extracting the amplitude coefficient G. These are transmitted over a link to the receiver where they are used to reconstruct the speech using a tube model. All these parameters are sampled at a rate of 50 Hz. If 4 bits are assigned to each of ten coefficents, 1 bit to voiced/unvoiced detection, 4 bits to pitch and 3 bits to G, then 48 bits can represent the information needed to operate the vocal cavity synthesizer. Sampling at a 50 Hz rate yields a channel bit rate of 2400 bps. Thus, the linear predictive vocoder is capable of transmitting the speech message at a low bit rate. However, when parity bits for error checking are added, the sampling rate should be reduced in order to accommodate the information bits and parity bits in a 2400 bps bit rate channel.

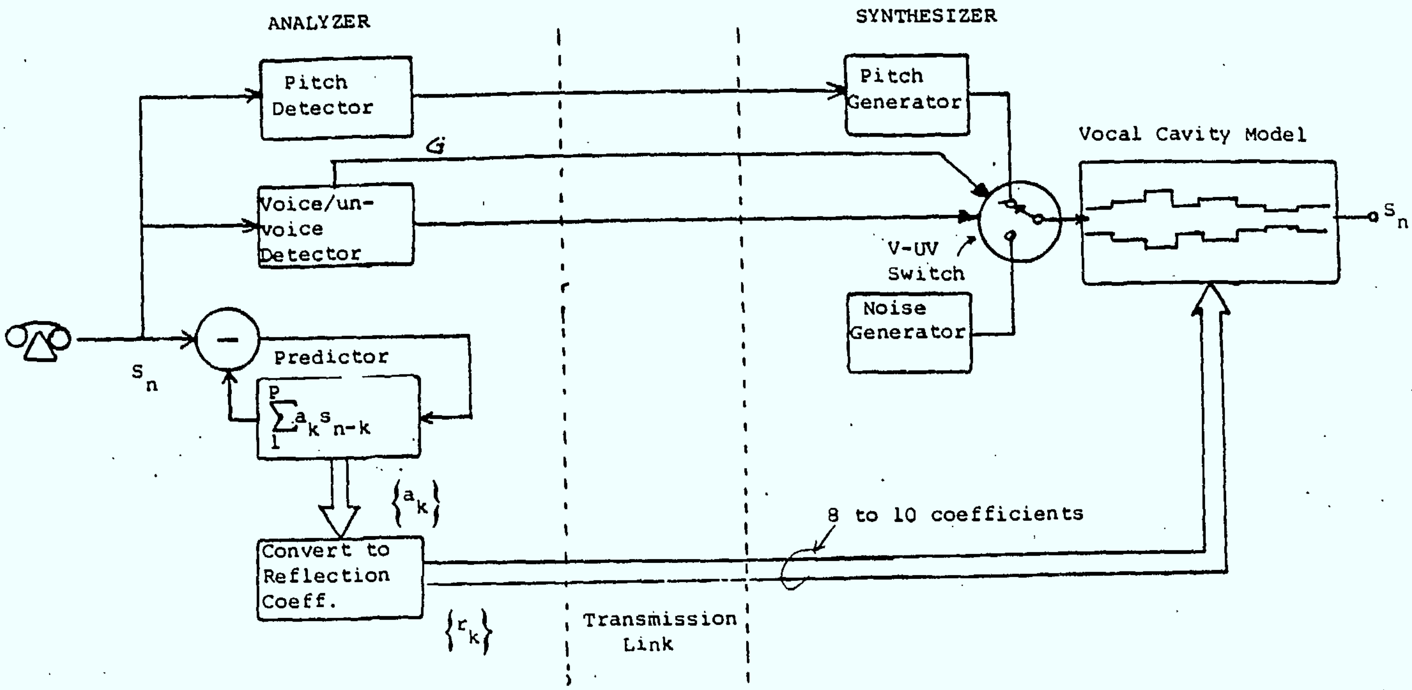

Figure D.3- Block diagram of a complete linear predictive vocoder.

2750Z/2751Z/2869Z

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APPENDIX E

PERFORMANCE SPECIFICATIONS FOR SOME AVAILABLE ACSSB TERRESTRIAL TERMINALS

Specifications for ACSSB radio manufactured by E.F. Johnson

Frequency of operation	150 - 174 MHz
Receiver @ 25 C:	
Sensitivity	0.25 mV for 12 dB SINAD (C-message)
Selectivity	
(Adj Channel)	65 dB @ +/-5 kHz spacing
Intermodulation	60 dB (3 generator method)
Image Rejection	60 dB
Spurious Response	60 dB
Pilot Acquisition Time	50 Millisecond maximum
Audio Distortion	
(Speech Path)	5% @ 1000 Hz at 100 UV
Audio Frequency Response	+2,-5 dB from a 6 dB/octave de-emphasis 300 Hz to 2900 Hz
Recv 1 dB compression	-40 dBm 50 ohm rf input
Radiated Spurious	FCC Rules - Part 15
Rx Carrier Stability	+/-2 ppm over voltage and temp unlocked condition
Transmitter @ 25 C:	
RF Power Output	18 Watt PEP @ Antenna Port
Carrier Level	10 dB below PEP
3rd Order Intermodulation	40 dB below PEP
5th Order Intermodulation	56 dB below PEP
Audio Response	+/-2,-5 dB from a 6 dB/octave pre-emphasis 300 Hz to 2900 Hz
Conducted Spurious	FCC Part-22 52 dB below PEP outside +/- 7.5 kHz
Radiated Spurious	Same as conducted
Carrier Stability	+/-2 ppm over voltage and temperature
General:	
Temperature Range *	-30 to +60 degrees Centigrade
Voltage Range *	13.8 VDC +/-10%

*Allowable degradation to be determined

Specifications for ACSSB radio by Contemporary Communications

Specifications

Transmitter -

RF Power Output	18 W. PEP at antenna port
Pilot Frequency	Suppressed Carrier
Sideband Used	Upper
Pilot Level	7 dB below PEP
3rd Order Intermod.	40 dB below PEP
5th Order Intermod.	56 dB below PEP
Audio Response	+2, -5 dB from a 6 dB/Oct. preemphasis 300 - 2900 Hz
Spurious Emissions-	
Conducted	70 dB below PEP
Radiated	63 dB below PEP
Carrier Stability	+/- 2ppm
Carrier Frequency	1500 Hz below channel center *

Receiver -

Sensitivity	0.25 uV./12 dB SINAD
Selectivity	65 dB at +/-5 kHz
Intermodulation	75 dB
Image Rejection	75 dB
Spurious Response	75 dB
Pilot Aquisition Time	50 msec. max.
Audio Distortion	5% at 1000 Hz
Audio Response	+2,-5 dB from a 6 dB/Oct. de-emphasis 300 - 2900 Hz
Radiated Spurious	Meets Part 15, FCC Rules

Other -

Temperature Range	-30 to + 60 degrees C.
Voltage Range	13.8 Vdc +/- 15%

* Frequency difference between carrier and channel center may be decreased.

Specifications for ACSSB terminals by Aerotron, Inc.

ACSB®
PIONEER 1000
VHF HIGH BAND
DASH MOUNT MOBILE TRANSCEIVER

PERFORMANCE SPECIFICATIONS

GENERAL

Frequency range (MHz)	136-174
Number of channels	4
Input voltage (nominal)	+13.6VDC Negative ground system
Input current at +13.6VDC	
Receiver, squelched (amps)	0.4
unsquelched (amps)	1.5
Transmitter, average (amps)	5
peak (amps)	7
Antenna switching	Solid State (no relays)
Power switching	Solid State (no relays)
Environmental	
Temperature (degrees)	-30C to +60C
Humidity (%)	0 to 100

TRANSMITTER

Power output (watts) PEP (Peak Envelope Power)	25
Output impedance (ohms)	50
Multi-frequency spread (MHz)	5
Frequency stability (± %)	0.0002 std.
Modulation	
Mode	3.6A3J with 3.1 KHz Identity Carrier
Distortion (%) maximum	3
Audio response	+1, −3dB from 300 to 2500Hz, 1000Hz reference
Spurious and harmonic output dB below rated PEP	60
FM hum and noise (dB)	-40

RECEIVER

Sensitivity	
20dB quieting (microvolts)	0.3
Input impedance (ohms)	50
Multi-frequency spread (MHz)	5
Frequency stability (± %)	0.0002 std.
Selectivity (dB)	
@ 5 KHz channel spacing	-60
@ 25 KHz channel spacing	-80
Intermodulation (dB)	-85
Spurious and image rejection (dB)	85
Audio power (watts) @ less than 5% distortion	10

PHYSICAL SPECIFICATIONS

Dimensions	9.5" W x 2.6" H x 11.0" D (24.1 cm W x 6.7 cm H x 23.0 cm D)
Weight	
Net	11.7 lbs. (5.3 kg)
Shipping	(15.0 lbs. (6.8 kg))

OPTIONS AND ACCESSORIES

- CTCSS single tone encoder/decoder
- CTCSS multiple tone encoders/decoder
- Microphone cable extender
- Palm microphone with built-in DTMF [illegible]
- 20 ft. power cable
- R.F. preamplifier
- Time out timer

Specifications subject to change without notice

2750Z/2751Z/2869Z

ACSB®

PIONEER 1000

VHF HIGH BAND

DESK-TOP BASE STATION

PERFORMANCE SPECIFICATIONS

GENERAL

Frequency range (MHz)	136-174
Number of channels	4
Input voltage (nominal)	120/240VAC 50/60Hz
Current Drain	
Receiver, squelched (Watts)	15
unsquelched (Watts)	65
Transmitter, average (Watts)	250
Antenna switching	Solid State (no relays)
Power switching	Solid State (no relays)
Environmental	
Temperature (degrees)	-30C to +60C
Humidity (%)	0 to 100

TRANSMITTER

Power output (watts) PEP (Peak Envelope Power)	25
Output impedance (ohms)	50
Multi-frequency spread (MHz)	5
Frequency stability (± %)	0.0002 std.
Modulation	
Mode	3.6A3J with 3.1 KHz Identity Carrier
Distortion (%) maximum	3
Audio response	+1, −3dB from 300 to 2500Hz, 1000Hz reference
Spurious and harmonic output dB below rated PEP	60
FM hum and noise (dB)	-40

RECEIVER

Sensitivity	
20dB quieting (microvolts)	0.3
Input impedance (ohms)	50
Multi-frequency spread (MHz)	5
Frequency stability (± %)	0.0002 std.
Selectivity (dB)	
@ 5 KHz channel spacing	-60
@ 25 KHz channel spacing	-80
Intermodulation (dB)	-85
Spurious and image rejection (dB)	85
Audio power (watts)	10 @ less than 5% distortion

PHYSICAL SPECIFICATIONS

Dimensions	11.4" W x 10.0" H x 17.0" D (28.3 cm W x 25.4 cm H x 43.2 cm D)
Weight	
Net	25 lbs. (11.3 kg)
Shipping	32 lbs. (14.5 kg)

OPTIONS AND ACCESSORIES

- Quartz digital clock (12 hr)
- Peak reading VU meter with light bar indicator
- CTCSS single tone encoder/decoder
- CTCSS multiple tone encoders/decoders
- R.F. preamplifier

Specifications subject to change without notice

2750Z/2751Z/2869Z

ACSB®

PIONEER 1000

VHF HIGH BAND

TRUNK MOUNT MOBILE TRANSCEIVER

PERFORMANCE SPECIFICATIONS

GENERAL

Frequency range (MHz)	136-174
Number of channels	4
Input voltage (nominal)	+13.6VDC Negative ground system
Input current at +13.6VDC	
Receiver, squelched (amps)	0.4
unsquelched (amps)	1.5
Transmitter, average (amps)	5
peak (amps)	7
Antenna switching	Solid State (no relays)
Power switching	Solid State (no relays)
Environmental	
Temperature (degrees)	-30C to +60C
Humidity (%)	0 to 100

TRANSMITTER

Power output (watts) PEP (Peak Envelope Power)	25
Output impedance (ohms)	50
Multi-frequency spread (MHz)	5
Frequency stability (± %)	0.0002 std.
Modulation	
Mode	3.6A3J with 3.1 KHz Identity Carrier
Distortion (%) maximum	3
Audio response	+1, −3dB from 300 to 2500Hz, 1000Hz reference
Spurious and harmonic output dB below rated PEP	60
FM hum and noise (dB)	-40

RECEIVER

Sensitivity	
20dB quieting (microvolts)	0.3
Input impedance (ohms)	50
Multi-frequency spread (MHz)	5
Frequency stability (± %)	0.0002 std.
Selectivity (dB)	
@ 5 KHz channel spacing	60
@ 25 KHz channel spacing	80
Intermodulation (dB)	-85
Spurious and image rejection (dB)	-85
Audio power (watts)	10 @ less than 5% distortion

PHYSICAL SPECIFICATIONS

Dimensions:

Transceiver	11.0" W x 3.1" H x 17.0" D (27.9 cm W x 7.9 cm H x 43.2 cm D)
Control head	5.0" W x 2.3" H x 1.6" D (12.7 cm W x 6.9 cm H x 4.1 cm D)
Speaker	5.0" W x 5.0" H x 2.5" (12.7 cm W x 12.7 cm H x 6.4 cm D)

Weight

Net	22.9 lbs. (10.4 kg)
Shipping	30.0 lbs. (13.6 kg)

OPTIONS AND ACCESSORIES

- CTCSS single tone encoder/decoder
- CTCSS multiple tone encoders/decoder
- Palm microphone with built-in DTMF [illegible]
- 25 ft. power cable
- R.F. preamplifier
- Time-out-timer

Specifications subject to change without [illegible]

2750Z/2751Z/2869Z

ACSB®
PIONEER 1000
VHF HIGH BAND
UP RIGHT BASE/REPEATER STATION

PERFORMANCE SPECIFICATIONS

GENERAL

Frequency range (MHz)	136-174
Number of channels	2
Input voltage (nominal)	120/240VAC 50/60Hz
Power Consumption	
Receiver, squelched (Watts)	90
unsquelched (Watts)	120
Transmitter, average (Watts)	320
Antenna switching	Solid State (no relays)
Power switching	Solid State (no relays)
Environmental	
Temperature (degrees)	-30C to +60C
Humidity (%)	0 to 100

TRANSMITTER

Power output (watts) PEP (Peak Envelope Power)	25
Output impedance (ohms)	50
Multi-frequency spread (MHz)	5
Frequency stability (± %)	0.0002
Modulation	
Mode	3.6A3J with 3.1 KHz Identity Carrier
Distortion (%) maximum	3
Audio response	+1, −3dB from 300 to 2500Hz, 1000Hz reference
Spurious and harmonic output dB below rated PEP	60
FM hum and noise (dB)	-40

RECEIVER

Sensitivity	
20dB quieting (microvolts)	0.3
Input impedance (ohms)	50
Multi-frequency spread (MHz)	5
Frequency stability (± %)	0.0002
Selectivity (dB)	
@ 5 KHz channel spacing	60
@ 25 KHz channel spacing	80
Intermodulation (dB)	85
Spurious and image rejection (dB)	35
Audio power (watts) @ less than 5% distortion	10

PHYSICAL SPECIFICATIONS

Dimensions	22" W x 41" H x 12"D (55.9 cm W x 104.1 cm H x 30.5 cm D)
Weight	
Net	90 lbs. (40.8 kg)
Shipping	130 lbs. (59.0 kg)

OPTIONS AND ACCESSORIES

- Single-tone CTCSS encoder/decoder
- Community repeater operation
- Dedicated repeater operation
- 120 VDC standby with automatic switchover and trickle charger
- DC or tone remote control
- Telephone interconnect
- Control/metering/test panel
- Full duplex operation
- Receive crystal filter

Specifications subject to change without notice

2750Z/2751Z/2869Z

SPECIFICATIONS FOR AIRFONE TERMINALS

EQUIPMENT DESCRIPTION

1. <u>Cordless Handset</u>

a) Model Number - Airfone Part No. AFA15-0100-AB

b) Frequency Range - Transmit - 49MHz Band) See Table 1 for Channel
Receive - 1.7MHz Band) & Frequency Assignment

c) Type of Modulation - FM direct; ±2.5kHz peak deviation

d) Frequency Stability of unmodulated transmitter carrier - ±0.0025% (25°C ref.)

e) Nominal Transmitter Power Output - 24mW into antenna

f) Spurious Transmitter Output Emissions - At least 35dB below 49MHz fundamental

g) Transmitter FM Noise Level - -50dB, 2.5kHz deviation at 1kHz reference

h) Receiver Frequency Stability - ±0.01% (25°C ref.)

i) Receiver RF Sensitivity - Adequate to achieve 20dB of quieting in an RF field of 50uV/m

j) Adjacent Channel Rejection - At least 50dB

k) Power Source - 5.2V battery pack, rechargeable when placed in cordless handset holder

l) Transmit Antenna - E-field device with no directivity nulls greater than 20dB at any orientation angle. It shall be housed in the handset and not protrude beyond the case by more than 5/8" with a maximum diameter of 5/8"

m) Receiver Antenna - H-field ferrite loop type with no directivity nulls greater than 30dB

2. <u>Cordless Handset Base Station - Part of airborne CMN</u>

a) Model Number - Airfone Part No. AFB10-0005-AA

b) Frequency Range - Transmit - 1.7MHz Band
Receive - 49 MHz Band

c) Type of Modulation - FM direct: ± 2.5kHz peak deviation

d) Frequency Stability of unmodulated transmitter carrier - ± 1.0kHz at 1.7MHz

e) Nominal Transmitter Power Output - +30dBm (1 Watt) ±1dB

f) Spurious Transmitter Output Emissions - Harmonic output shall not exceed -20dB. Spurious output shall not exceed -50dB

g) Transmitter FM Noise Level - -50dB, reference 2.5kHz deviation with a 1kHz tone

h) Receiver RF Sensitivity - 20dB quieting with a 1.2uV RF signal input at 50 ohms. 30dB with a 10uV RF signal input

i) Receiver Deviation Acceptance - ±2.0 kHz at 10% Distortion

j) Receiver Selectivity - 55.0dB

k) Receiver Image and Spurious Response - 55.0dB

l) Receiver AM Rejection - 50.0 dB

m) Power Source - 24.0V DC, ±20%, negative ground

n) Ceiling Antenna - 300 ohm antenna wire meeting the requirements for installation in an aircraft. It is fed via a hybrid power combiner for transmit and it feeds a power divider for receive.

3. <u>Airborne Transceiver</u>

a) Manufacturer - E.F. Johnson
Model Number - 242-0747-6000
Airfone Part No. - AF05-0100-AA

b) Frequency Range - Transmit - 899 to 901MHz
Receive - 944 to 946MHz - See Table 2 for Receive Channel & Frequency Assignment

c) Type of Modulation - Compandored and compatible single sideband, lower sideband

d) Maximum Percentage Modulation - 100%

e) Necessary Bandwidth - 6kHz for each sub-channel

f) Frequency Stability of unmodulated transmitter carrier - $\pm$ 100Hz

g) Nominal Transmitter Power Output - 8 Watts, +2/-1 Watts

h) Spurious Transmitter Output Emissions - Attenuated by at least 43 + 10 $\log_{10}$ (Carrier Power Output), in dB

i) Receiver Bandwidth - Adjacent channel selectivity at least 60dB, measured by the EIA method for AM receivers

j) Receiver Image Rejection Ratio and Spurious Response - At least 85dB including 1/2 IF rejection. Image rejection at least 85dB

k) IF Frequency - 17.15MHz, +1MHz, with local oscillator tuned below the receiver carrier frequency

l) 2nd IF Bandwidth - 200kHz to -3dB points

m) Receiver Frequency Stability - $\pm$100Hz, crystal controlled

n) Sensitivity - For 10dB SINAD, better than -116dBm for an AM signal centered in the first IF filter, modulated 80%

o) AGC - Over the input range of 0.5uV to 1mV for a 30% AM signal, the audio output shall not change more than $\pm$1dB

p) FM Section

i) Sensitivity - Better than 0.5uV and 600Hz deviation for a 10dB SINAD

ii) AGC - Over the range of 0.5uV to 1mV when viewing a 1kHz tone deviated 1,500Hz, the audio output shall not change more than 2dB

iii) Adjacent Channel Selectivity - The FM receiver sensitivity shall not be degraded by more than 3dB by an 80% modulated AM signal $\pm$6kHz removed from the FM carrier and at a level of 20dB above the FM signal over a range of FM signals from -100dBm to -50dBm input level

iv) Spurious Response Attenuation - 80dB

v) Hum and Noise - -60dB

vi) AFC - Correct transmit frequency $\pm$1kHz, minimum, for Doppler shift correction. The correction must be accurate within $\pm$50Hz

vii) Data Reception - Minimum bit error rate 1 in 1 million with standard audio tones of 1170Hz and 2225Hz Standard ASCII at 300 baud. The FM receive shall pass FSK data at 1070 and 1270Hz with less than 7% distortion and less than 8dB of twist

p) Antenna

There are two (2) antennas per aircraft equipped with four (4) cabin telephones. One antenna is used for receive and two (2) transmitters, while the second is used for the two (2) added transmitters. There are separate combiner/splitters and filters in each of the transmit and receive feeders

i) Manufacturer - Sensor Systems Inc.
Model Number - S65-5366-901L

ii) Horizontal Directivity Pattern - Omni (See Figures 1-A to 1-C) Gain - Approximately the same as 1/4 wavelength monopole, 5dBi maximum

q) 900MHz Cavity Filter

i) Model Number - Airfone Part No. AFP-0002-AA

ii) VSWR - 1.25:1 at 899MHz, and 901MHz

iii) Insertion Loss - Less than 1.0dB at above frequencies

iv) 3.0dB Bandwidth - 4.0MHz centered on 900MHz

v) Minimum Attenuation at 944MHz - 50dB

vi) Minimum Attenuation at 866.7MHz - 45dB

r) 945MHz Cavity Filter
   i) Model Number - Airfone Part No. AFPO5-0003-AA
   ii) VSWR - 1.25:1 at 944MHz, 945MHz and 946MHz
   iii) Insertion Loss - Less than 1.0dB at the above frequencies
   iv) 3.0dB Bandwidth - Centered at 945MHz
   v) Minimum Attenuation at 901MHz - 50dB
   vi) Minimum Attenuation at 911.7MHz - 45dB
   vii) Minimum VSWR at 901MHz - 50 to 1

4. <u>Ground Transceiver</u>

a) Manufacturer - E.F. Johnson
   Model Number - 242-0747-700
   Airfone Part No. - AFA05-1006-GS

b) Frequency Range - Transmit - 944 to 946MHz
   Receive - 899 to 901MHz - See Table 1 for Transmit Channel & Frequency Assignment

c) Type of Modulation - Compandored and compatible single sideband, lower sideband

d) Maximum Percentage Modulation - 100%

e) Necessary Transmit Bandwidth - 4kHz for each sub-channel

f) Frequency Stability of unmodulated carrier - ±100Hz

g) Nominal Transmitter Power Output - 8 Watts, ±1dB

h) Spurious Emissions - Attenuated by at least $43 + 10 \log_{10}$ (Mean power output), in dB

i) Upper sideband Rejection - At least 25dB with respect to lower sideband

j) Receiver Sensitivity - For 10dB SINAD, better than -113dBm for an AM signal centered in the last IF filter, modulated at 80%

k) Receiver AGC - Audio output level change shall be less than ±1dB over the range of -107 to -47dBm for a 30% modulated input signal

l) Receiver Adjacent Channel Selectivity - At least 40dB as measured by EIA RS-382

m) Receiver Spurious Response Attenuation - At least 85dB including 1/2 IF rejection. Image frequencies shall also be rejected a minimum of 85dB when measured with a down converter and a 900MHz Lorch filter

n) IF Frequency - 17.15MHz, ±1MHz, with local oscillator tuned below the receiver carrier frequency

o) 2nd IF Bandwidth - 200 kHz at -3dB points

p) Receiver Frequency Stability - ±100Hz, crystal controlled

q) Receiver Hum and Noise - At least -45dB

r) Pilot Transmitter
   i) Manufacturer - E.F. Johnson Company
      Model Number - 242-0747-400
   ii) Frequency Range - 944 to 946MHz, pre-set to one channel in that band
   iii) Type of Modulation - Phase modulation, ± 1 radian over the range of 300 to 4,000Hz
   iv) Maximum Modulation - ±1 radian, ±0.1 radians for an audio input of 0.3V rms
   v) Necessary Bandwidth - 6kHz
   vi) Frequency Stability of unmodulated carrier - ±100Hz, controlled by oven controlled crystal oscillator
   vii) Nominal Power Output - 15 watts, minimum
   viii) Level of Spurious and Harmonic Emissions - Spurious emissions shall meet EIA RS-152B. Harmonic emissions shall be at least 80dB below fundamental

s) Antenna - Separate antennas are used for receive and for each of the transmitters. There are four (4) antennas per pole, that is three transmit and one receive.

   i) Manufacturer - Decibel (db) Products
      Model Number - DB803Q-M
      Airfone Part Number - AFP25-0002-GS
      Frequency Range - 899 to 901MHz and 944 to 946MHz
   ii) Horizontal Pattern - Omni (See Figures 2-A & 2-B) Vertical Pattern Beamwidth - 46° to -3dB points
   iii) Gain - 3dB (min) over half wave dipole
   iv) VSWR - 1.5:1 or less

t) 900MHz Cavity Filter

i) Model Number - Airfone Part No. AFPO5-0002-AA
ii) VSWR - 1.25:1 at 899MHz, 900MHz and 901MHz
iii) Insertion Loss - Less than 1.0dB at above frequencies
iv) 3.0dB Bandwidth - 4.0MHz centered on 900MHz
v) Minimum Attenuation at 944MHz - 50dB
vi) Minimum Attenuation at 866.7MHz - 45dB

u) 945MHz Cavity Filter

i) Model Number - Airfone Part No. AFPO5-0003-AA
ii) VSWR - 1.25:1 at 944MHz, 945MHz and 946MHz
iii) Insertion Loss - Less than 1.0dB at above frequencies
iv) 3.0dB Bandwidth - Centered on 945MHz
v) Minimum Attenuation at 901MHz - 50dB
vi) Minimum Attenuation at 911.7MHz - 45dB
vii) Minimum VSWR at 901MHz - 50 to 1