Theory and Implementation of a Splitband Modem Using the TMS32010

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Theory and Implementation of a Splitband Modem Using the TMS32010

Abstract

This report discusses the theory and implementation of a 1200-bps splitband modem, the Bell 212A/V.22. Throughout the report, the terms Bell 212A and V.22 both refer to implemented modem.

The report covers the following topics:

- Modem Functional Blocks
  - Modem Transmitter
  - Modem Receiver
- Modem Hardware Description
  - Analog Interface
  - Filters
  - Data Converters
- Functions Implemented in the TMS32010
  - Transmit Filters
  - Receive Filters
  - AGC Implementation
  - Carrier Recovery Implementation
  - Baud Clock Alignment Implementation

- Functions Implemented in the TMS7742
  - Asynchronous-to-Synchronous and Synchronous-to-Asynchronous Conversions
  - Scrambler/Descrambler
- Performance
- Other Implementation Considerations
- Conclusions and References

The report also includes the following appendixes:

- Appendix A Derivation of DM Structure Equations
- Appendix B Effects of Nonideal Hilbert Transformers
- Appendix C AGC Table Generator Code
- Appendix D TMS32010 Source Code
- Appendix E TMS7742 Source Code

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Introduction

With the predominant usage of computers and especially PCs, data communications are of increasing importance. Communication between the various computer systems and terminals is frequently accomplished by means of the Public Switched Telephone Network (PSTN). The essential element for this data communication is the modem, which interfaces computer systems and terminals with the telephone network.

In the past, modems have been traditionally implemented in the analog domain using discrete components. Recently, modem manufacturers have realized the flexibility and high performance offered by digital approaches. With the drastic reduction in the cost of digital signal processors, the power of Digital Signal Processing (DSP) becomes available for the implementation of medium-speed and high-speed modems.

This application report discusses the digital implementation of a modem using the TMS32010 Digital Signal Processor. Attention is focused on splitband modems, a class of modems that splits the bandwidth of the communications channel (telephone network) so that full-duplex operation can occur. The splitband technique is mainly used for implementing modems with data rates up to 2400 bps (bits per second). This report describes the theory and implementation of the Bell 212A/V.22 Recommendation, a 1200-bps splitband modem. Note that in the remainder of this report, the designations Bell 212A and V.22 are used synonymously to refer to the modem implemented. This report is not intended to provide a commercial product, but to introduce the implementation considerations and merits of digital signal processing-based approaches. Some of the protocol requirements for the Bell 212A/V.22 Recommendation are not implemented: the answer mode, the 300-bps Frequency Shift Keying (FSK) modem, and the notch filter required to reject the guard tone from the received signal.

Modems are sophisticated devices consisting of many functional blocks that must be correctly implemented. The interface of the functional blocks must also be appropriately adjusted for the overall structure to function properly. The different functional blocks can be implemented in many ways. For example, the receiver input bandpass filters can be recursive or nonrecursive, and different algorithms can be used for the carrier recovery and clock recovery. In addition to the possibility of implementing different algorithms, new algorithms may need to be added to the already existing structure, such as an adaptive equalizer or a second loop within the carrier recovery for the suppression of carrier-phase jitter. These considerations indicate the advantage of the microprocessor-based over the analog-based approach. Using the microprocessor approach, the designer can test different algorithms by simply modifying the software. Additional functional blocks can be included by simply adding new code. Therefore, high-performance modems can be implemented in a very short period of time.

The computational burden for digital modem implementation is very heavy. This implies the need of special features for the microprocessor to be used. The TMS32010, with its 200-ns cycle time, on-chip multiplier, and specialized instruction set is uniquely architected for digital signal processing. Because of this, the TMS32010 can implement the modem functional blocks using only a portion of its available processing power. Another major advantage of the microprocessor approach is the possibility of implementing variable-rate modems using the same hardware. Specifically, the same hardware used for the implementation of the Bell 212A/V.22 Recommendation can be used to implement 2400-bps splitband modems (CCITT V.22 bis Recommendation) by merely changing the software. Besides implementing various modems using the same hardware, additional functions can be included, such as speech store-and-forward and the Data Encryption Standard (DES)[1] for secure data communications.

This report is organized as follows: The first section, Modem Functional Blocks, is a description of the functional blocks required for implementation of the Bell 212A/V.22 Recommendation. The second section, Modem Hardware Description, is a brief discussion of the hardware used for the modem implementation. The functions implemented within the TMS32010 are described in detail in the third section, while the fourth section contains an overview of the functions implemented in the modem controller (the Texas Instruments TMS7742 microcomputer). The performance of the TMS32010-based modem is presented in the fifth section. Finally, the last section suggests alternative hardware configurations that can result in reduced system cost.

The prerequisites for understanding and gaining maximum benefit from this report are the level of a Bachelor's degree in Electrical Engineering and a basic understanding of Digital Signal Processing and Data Communications. Background material can be found in *Digital Signal Processing* (Chapters 1 through 7) by A.V. Oppenheim and R.W. Schafer; "Implementation of FIR/IIR Filters with the TMS32010/TMS32020," an application report in the book, *Digital Signal Processing Applications with the TMS320 Family*, offered by Texas Instruments; and in *Understanding Communications Systems* and *Understanding Data Communications*, books published by Texas Instruments.

Modem Functional Blocks

A modem (MOdulator-DEModulator) is a device that modulates the baseband information at the transmitter, and demodulates the received signal to retrieve the baseband information at the receiver. The Bell 212A is a full-duplex modem with the receiver and transmitter sharing the available bandwidth of the communications channel. This type of modem is said to operate in either the originate or answer mode (see Figure 1). In the originate mode, it initiates the communication process, transmits with a carrier frequency of 1200 Hz, and receives at the frequency of 2400 Hz. At the other end of the communications channel is a modem that operates in the answer mode, i.e., receives at 1200 Hz and transmits at 2400 Hz.

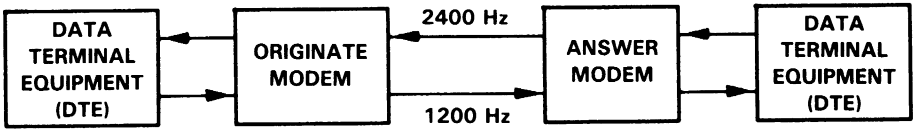


Figure 1. Originate/Answer Configuration

Table 1 shows the different functional blocks that comprize the modem transmitter and receiver.

Table 1. Modem Functional Blocks

Modem Transmitter	Implemented
Guard Tone Generator	No
Scrambler	Yes
Encoder	Yes
Digital Lowpass Filters	Yes
Originate Mode Modulator	Yes
Answer Mode Modulator	No
Modem Receiver	**Implemented**
Notch Filter	No
Originate Mode Bandpass Filters	Yes
Answer Mode Bandpass Filters	No
Automatic Gain Control	Yes
Demodulator	Yes
Decision Block	Yes
Decoder	Yes
Descrambler	Yes
Clock Recovery	Yes
Carrier Recovery	Yes

In the following two subsections, the operation of the modem transmitter and receiver are described. The transmitter accepts data (bits) from the Data Terminal Equipment (DTE). The DTE may be a dumb terminal, a PC, or a mainframe computer. The modem transmitter then performs the necessary processing to place this data in the proper form for transmission through the Public Switched Telephone Network. This processing basically consists of the modulation of the baseband information (logical ones and zeros (bits) sent by the DTE) into the passband of the communications channel for transmission. The receiver collects the information from the telephone network and transforms it back to its original form, i.e., the bits sent by the DTE.

Modem Transmitter

The Bell 212A/V.22 is a 1200-bps modem that uses the Differential Phase Shift Keying (DPSK) modulation technique to transmit data through the communications channel. In the first part of this subsection, the equation that describes the operation of Differential Phase Shift Keying modulation systems is derived from an intuitive approach. A rigorous derivation is given in Appendix A. The rest of this subsection discusses the functional blocks required to correctly implement this equation.

In Differential Phase Shift Keying, the information is encoded as the phase change of the transmitter carrier. With $\phi(n)$ denoting the phase that contains the information to be transmitted, the transmitted signal s(n) is represented mathematically by

$$s(n) = A \cos(\omega n + \phi(n)) \tag{1}$$

where ω is the carrier frequency. The parameter A determines the amplitude of the transmitted signal. Use of the trigonometric identity

$$\cos(X+Y) = \cos(X)\cos(Y) - \sin(X)\sin(Y)$$

gives

$$s(n) = A\,\{\cos(\omega n)\cos[\phi(n)] - \sin(\omega n)\sin[\phi(n)]\} \tag{2}$$

The substitution of

$$I(n) = A\cos[\phi(n)]$$

$$Q(n) = -A\sin[\phi(n)]$$

into (2) results in (3) used to describe DPSK modulation systems.

$$s(n) = I(n)\cos(\omega n) + Q(n)\sin(\omega n) \tag{3}$$

From (3) it can be seen that the transmission of the baseband sequence $\{I(n),Q(n)\}$ is accomplished by using two separate modulation carriers, a sine wave and a cosine wave. These waves are orthogonal; i.e., the information in the direction of the one wave (cosine) is independent of the information in the direction of the other wave (sine), and therefore this information is recoverable. Each value of the $\{I(n),Q(n)\}$ sequence corresponds to one signaling element (symbol) transmitted. The number of signaling elements transmitted per second is commonly referred to as the baud rate, which for the Bell 212A/V.22 is set by the protocol to 600.

Some widely used terminology becomes apparent from (3). The baseband sequence that modulates the cosine wave is called the In-phase sequence. The baseband sequence that modulates the sine wave is called the Quadrature-phase sequence since the sine-wave carrier is 90 degrees (one Quadrant) out-of-phase from the cosine-wave carrier. The part of the transmitter/receiver that processes the In-phase sequence is commonly referred to as the I-channel, while the part of the transmitter/receiver that processes the Quadrature-phase sequence is referred to as the Q-channel.

The derivation of (3) indicates that the incoming sequence $d_s(n)$ is encoded into the sequence $\{I(n),Q(n)\}$, and the latter is transmitted. The mapping rule used is unique for each system; i.e., the mapping rule used for the Bell 212A/V.22 is different from the mapping rules used for other modems (V.22 bis, V.27, V.29, etc.). For example, for the Bell 212A/V.22, the sequence $\{I(n),Q(n)\}$ contains phase information, while for the V.22 bis, it contains phase and amplitude information. The set of possible values of the sequence $\{I(n),Q(n)\}$ determines the signal constellation, which is given in a two-dimensional representation.[2] The signal constellation, commonly referred to as the constellation diagram, is a geometric picture that emphasizes the fact that the two channels are 90 degrees (Quadrature) out-of-phase.

The Bell 212A/V.22, with a 600-baud rate, accomplishes the transmission of 1200 bps by encoding two incoming bits (dibit) in a single baud. Since there are four possible values for every dibit, the constellation diagram for the Bell 212A/V.22 contains four points. Each constellation point, i.e., each value of the $\{I(n),Q(n)\}$ sequence, corresponds to a total phase value to be transmitted. The calculation of the total phase from the incoming dibits will be discussed later. Figure 2 shows the constellation diagram for the Bell 212A/V.22. The four constellation points, notated A, B, C, and D, lie on a circle. Since there is no amplitude information transmitted, the radius of this circle is normalized to unity. The total phase information represented by each constellation point is enclosed in parentheses.

The encoding of the incoming sequence $d_s(n)$ into the values of the sequence $\{I(n),Q(n)\}$ is implemented by the encoder. The encoder is a one-input, two-output functional block, whose function is to map every two incoming bits (dibit) of the incoming sequence $d_s(n)$ to a total phase. The total phase is then represented by the values of the sequence $\{I(n),Q(n)\}$, and the latter is transmitted. The mapping rule used to encode the total phase into the values of the coder outputs $\{I(n),Q(n)\}$ is shown in Table 2. Each $\{I,Q\}$ entry in this table corresponds to one point in the constellation diagram of Figure 2. This is indicated in the third column of Table 2.

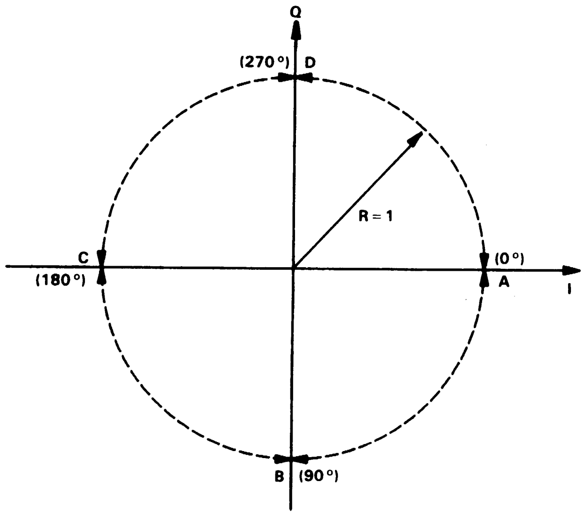


Figure 2. Signal Constellation for the Bell 212A/V.22

Table 2. Total-Phase-to-Coder-Output Mapping Rule

Total Phase	Encoder Output { I , Q }	Point in Constellation Diagram of Figure 2
0 degrees	{ 1 , 0 }	A
90 degrees	{ 0 , −1 }	B
180 degress	{−1 , 0 }	C
270 degrees	{ 0 , 1 }	D

The calculation of the total phase from the incoming dibits is accomplished in two steps. First, each incoming dibit is mapped to a unique phase change. Second, this phase change is added to the previous total phase to obtain the new total phase. The mapping rule used to uniquely map each dibit to a phase change is shown in Table 3.

Table 3. Dibit-to-Phase Change Correspondence

Dibit	Phase Change
00	90 degrees
01	0 degrees
10	180 degrees
11	270 degrees

To illustrate the two-step procedure used to calculate the total phase, consider the following example. The previous total phase is 90 degrees, and the incoming dibit is 10. From Table 3, the phase change corresponding to a 10 dibit is 180 degrees. Therefore, the new total phase is

$$\text{new total phase} = \text{modulo } 360 \text{ (previous total phase + phase change)}$$

$$= (90 \text{ degrees}) + (180 \text{ degrees}) = 270 \text{ degrees}$$

Using Table 2, for this value of total phase (270 degrees), the encoder output is $\{I,Q\} = \{0,1\}$. For the next incoming dibit, the above procedure is repeated with a 270-degree previous total phase.

At the receiver, the total phase is determined from the received $\{I,Q\}$ value. This total phase is subtracted from the previous total phase (the one transmitted during the previous baud), and the difference is the phase change. Since the phase-change-to-dibit mapping is unique, using the calculated value of the phase change results in the transmitted dibit being uniquely recovered at the receiver.

This differential approach (i.e., the calculation of the phase change instead of an absolute phase) is used because if the dibits were to correspond to an absolute phase, then a common-phase reference for both the receiver and the transmitter would be required. This in turn implies the need of a training sequence between the transmitter and the receiver so that a common-phase reference can be established. This training sequence, however, is not provided for the Bell 212A/V.22.

An overall block diagram for the modem transmitter is shown in Figure 3.[3] The basic structural blocks are the scrambler, encoder, digital lowpass filter, and digital modulator.

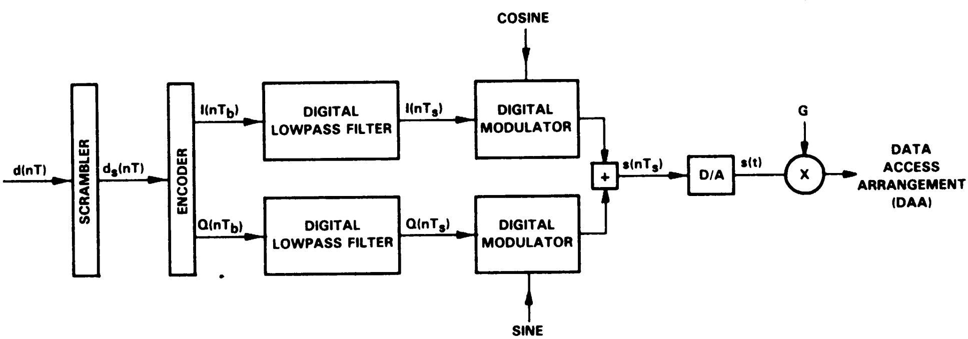


Figure 3. Modem Transmitter Block Diagram

Scrambler

The scrambler scrambles the bits sent by the DTE. To understand the need for a scrambler, consider the situation where the DTE sends a series of 01 dibits. From Table 3, each 1 dibit corresponds to a 0-degree phase change. Therefore, the total phase transmitted is the same. From the geometrical point of view, this results in transmitting the same constellation point (same total phase). At the receiver end, however, phase changes are required for correct clock recovery (see the clock recovery discussion in the Modem Receiver subsection). Therefore, the transmission of a series of 01 dibits generates problems for the receiving modem, such as losing carrier lock. To avoid this, the scrambler is introduced to minimize the probability that such 'ill-conditioned' dibits occur. With d(nT) input to the scrambler, the output $d_S(nT)$ is given by

$$d_S(nT) = d(nT) \text{ XOR } d_S((n-14)T) \text{ XOR } d_S((n-17)T) \tag{4}$$

where XOR indicates the exclusive-OR operation and T is the data period, i.e., the time between two successive bits sent by the Data Terminal Equipment. The signal flowgraph of the modem transmitter scrambler is shown in Figure 4 in which z^{-n} is used to indicated an n-sample delay.

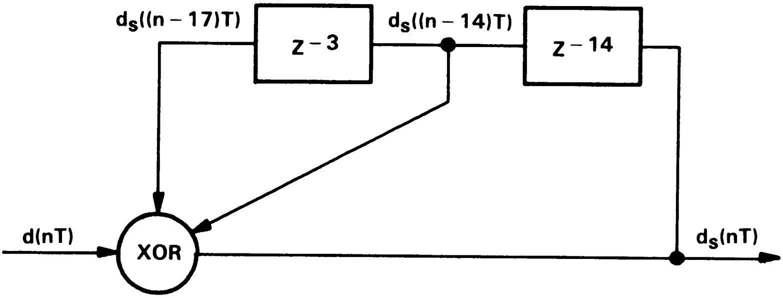


Figure 4. Signal Flowgraph of Transmitter Scrambler

Encoder

The function of the encoder, i.e., the mapping of the incoming sequence $d_S(n)$ to the values of the sequence $\{I(n),Q(n)\}$, was discussed earlier. However, there is one more related issue associated with the encoder, i.e., the change of the sampling frequency at the encoder output. Every two bits that the modem transmitter accepts from the DTE correspond to a unique phase to be transmitted. Therefore, at the encoder output, the sampling period changes from T (sampling period of incoming data) to T_b, i.e., from 1/1200 s to 1/600 s. The subscript b in T_b represents baud since the encoder output (I and Q channels) changes at the baud rate. The above discussion implies that $T_b = 2T$; i.e., the I and Q channels are updated after every pair of bits received from the DTE.

Digital Modulators and Lowpass Filters

Since the telephone network behaves as a bandpass filter with the passband starting around 300 Hz and ending around 3200 Hz, the baseband encoder outputs, $I(nT_b)$ and $Q(nT_b)$, cannot be directly transmitted through the communications medium. They first must be modulated up in frequency. The modulation is not attempted directly on the encoder outputs for two reasons. First, as discussed at the end of this subsection, the sampling frequency must increase from $1/T_b$ to $1/T_s$, with $1/T_s$ being at least 6.4 kHz. This increase in the sampling frequency is accomplished by interpolation. Second, if the modulation is attempted directly on the encoder outputs, the instantaneous changes of the $I(nT_b)$ and $Q(nT_b)$ generate higher-order harmonics. Some of these harmonics fall in the frequency region reserved for the receiver. To eliminate the harmonics and to also increase the sampling frequency by interpolation, the encoder outputs must be digitally lowpass-filtered. The characteristics and the implementation of the digital lowpass filters are discussed in detail in the Transmit Filters subsection of ''Functions Implemented in the TMS32010.''

At the output of the lowpass filters, the I-channel modulates a cosine wave and the Q-channel a sine wave. The modulating frequency is 1200 Hz for an originate modem and 2400 Hz for a answer modem. Finally, the two channels are summed before they are transformed into the analog signal transmitted through the telephone network. The output of the digital transmitter (before the D/A converter) is given by equation (3), repeated below for convenience.

$$s(nT_s) = I(nT_s) \cos(\omega nT_s) + Q(nT_s) \sin(\omega nT_s)$$

The sampling period T_s is $T_s = 1/f_s$ where f_s is the sampling frequency. This frequency must be at least twice the highest frequency component of the transmitted information (Nyquist rate) to satisfy the sampling theorem. Since the telephone network cuts off at approximately 3.2 kHz, the sampling frequency must be at least 6.4 kHz. Practical considerations (integer number of samples per baud, etc.) impose the necessity of higher sampling rates. For the present application, the sampling frequency used was 9.6 kHz. Since the baud frequency is 600 Hz, 16 (9600/600) samples correspond to each baud interval.

Modem Receiver

This subsection discusses the issues associated with the functional blocks required to implement a Bell 212A/V.22 modem receiver. The receiver structure is more sophisticated than that of the transmitter. For a low bit-error-rate performance (percentage of error bits received), an Automatic Gain Control (AGC) subsystem, adaptive equalization of the overall transmitting system, and a noise-independent carrier recovery and clock recovery are required. Since the issues associated with the carrier recovery and the clock recovery are critical in a modem design and difficult to understand, a good portion of this subsection is devoted to their discussion.

The adaptive equalizer is an adaptive filter that compensates for intersymbol interference and Doppler spread effects introduced during transmission over the telephone lines.[4] The magnitude of these effects depends on the bit rate and the quality of the telephone line. The effects are more severe at high bit rates (2400 bps and above) and over a worst-case telephone line, which is commonly represented by the 3002 line simulator.[5] The Bell 212A/V.22 protocol does not require the presence of an adaptive equalizer; therefore, this implementation does not include one. However, for increased performance on a 3002 line where even at medium speeds, such as 1200 bps, intersymbol interference and Doppler spread effects become severe, an adaptive equalizer is recommended. An important point here is that the addition of an adaptive equalizer in the current TMS32010 implementation of the Bell 212A/V.22 modem does not require any hardware changes. Increased performance results from an increase in the algorithmic sophistication.

An overall block diagram of the modem receiver is shown in Figure 5. The basic structural blocks of the modem receiver are the input bandpass filters, the automatic gain control (AGC), the demodulator, the decision block, the decoder, the descrambler, the carrier recovery, and the clock recovery.

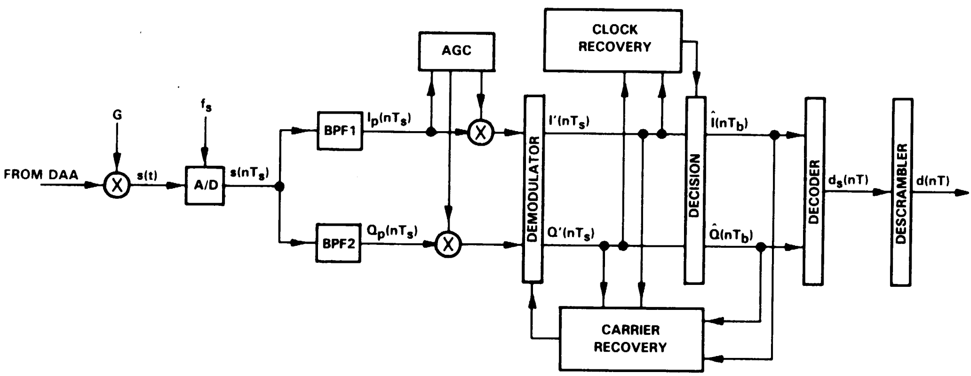


Figure 5. Modem Receiver Block Diagram

Input Bandpass Filters

The incoming analog signal s(t) is digitized at the sampling frequency f_S to obtain its digital counterpart $s(nT_S)$. This signal is then bandpass-filtered for three reasons:

1. Rejection of out-of-band noise, including the rejection of the transmit signal spectrum due to the near-end echo path,
2. Introduction of 90-degree relative phase shift required for the I and Q channel separation (see Appendix A), and
3. Fixed equalization for line distortion.

The second reason mentioned above implies the need of receiver bandpass filters that achieve a 90-degree relative phase shift. It is theoretically justified in Appendix B that if the two bandpass filters, denoted by BPF 1 and BPF 2 in Figure 5, achieve an exact 90-degree relative phase shift, there are no harmonics at the output of the receiver demodulator. If this condition is not met, harmonics appear at twice the carrier frequency. These harmonics were observed in the modem implementation when a set of bandpass filters not meeting the above condition was used. Elimination of the harmonics due to an inexact 90-degree relative phase shift involves the use of lowpass filters at the output of the demodulator. However, the group delay and the possible phase distortion introduced by the lowpass filters affect the carrier recovery and decision algorithms. Compensation for these side-effects of the lowpass filters results in a more complicated modem receiver design.

In the analog domain, where component drift is due to aging and/or temperature, it is virtually impossible to design bandpass filters or Hilbert transformers that achieve an exact 90-degree relative phase shift. Hilbert transformers, a special class of filters, are discussed in Appendices A and B. In the digital domain, however, the design of bandpass filters or Hilbert transformers that achieve an exact 90-degree relative phase shift is relatively easy. Digital filter design packages, such as the Digital Filter Design Package (DFDP) offered by the Atlanta Signal Processors Incorporated (ASPI), can be used to design modem receiver input filters on the TMS32010 that meet the exact 90-degree relative phase shift requirement. The characteristics and implementation of the modem receiver input bandpass filters are discussed in detail in the Receive Filters subsection of "Functions Implemented in the TMS32010."

Automatic Gain Control (AGC)

Because of the attenuation introduced by the telephone lines, the peak-to-peak voltage of the incoming analog signal s(t) ranges between 2 mV and 700 mV. However, signal levels in the receiver must be independent of the attenuation introduced by the communications channel. This is of paramount importance because the carrier recovery and clock recovery algorithms use error signals and thresholds dependent on the I and Q channel values. Therefore, the Automatic Gain Control subsystem is required to adjust the envelope of the I and Q channels so that they are of the same magnitude. The AGC algorithm used and its implementation is discussed in the Automatic Gain Control Implementation subsection of "Functions Implemented in the TMS32010."

Demodulator

The demodulator translates the passband information back to the baseband. With $I_p(nT_S)$ and $Q_p(nT_S)$ inputs to the demodulator (see Figure 5), the outputs $I'(nT_S)$ and $Q'(nT_S)$ are given by (see derivation in Appendix A)

$$I'(nT_S) = I_p(nT_S)\cos(\omega' nT_S) + Q_p(nT_S)\sin(\omega' nT_S) \tag{5}$$

$$Q'(nT_S) = I_p(nT_S)\sin(\omega' nT_S) - Q_p(nT_S)\cos(\omega' nT_S) \tag{6}$$

where ω' is the local carrier frequency. Figure 6 shows the demodulator structure that implements (5) and (6).

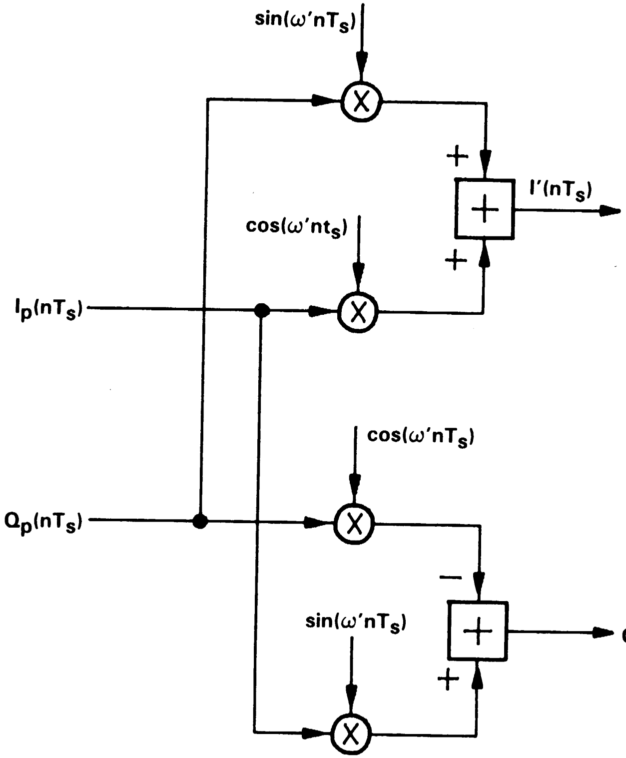


Figure 6. Demodulator Structure

Even with an ideal receiver, the $I'(nT_S)$ and $Q'(nT_S)$ channels shown in Figure 6 are 'noisy' replicas of the baseband I and Q channels at the output of the transmitter digital lowpass filters. The 'noise' has been injected by the telephone network as group delay, frequency jitter, and Gaussian noise.[4]

Decision Block and Descrambler

The decision block in Figure 5 calculates the total phase from the values of the baseband I and Q channels. By subtracting it from the previous total phase (the phase transmitted during the previous baud interval), the phase change is computed. Each phase change (total of four) has a corresponding unique dibit (see Modem Transmitter subsection). This dibit is fed into the descrambler (see Figure 5) to recover the originally transmitted dibit. The output of the descrambler is described by

$$d(nT) = d_S(nT) \text{ XOR } d_S((n-14)T) \text{ XOR } d_S((n-17)T) \qquad (7)$$

where T is the data period (1/1200 s for the Bell 212A). The signal flowgraph of the receiver descrambler is shown in Figure 7.

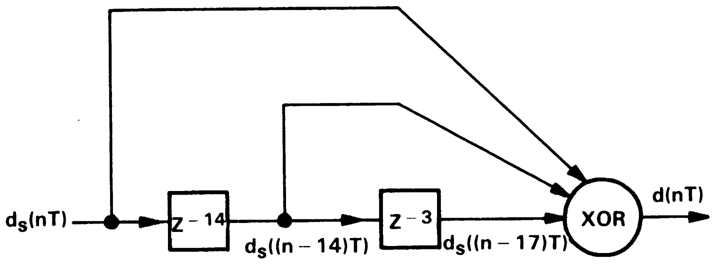


Figure 7. Signal Flowgraph of Receiver Descrambler

Carrier Recovery

A very important task of the modem receiver is the generation of a carrier that has the same frequency and phase with the incoming carrier. This receiver-generated carrier, called the local carrier, is used by the demodulator of Figure 6 to demodulate the incoming signal and therefore retrieve the baseband information. The process of generating this carrier is called carrier recovery. The standard approach to this is to use a phase-locked loop.[6] Figure 8 shows the basic blocks of a phase-locked loop: the phase detector (PD), loop filter and Voltage Controlled Oscillator (VCO).

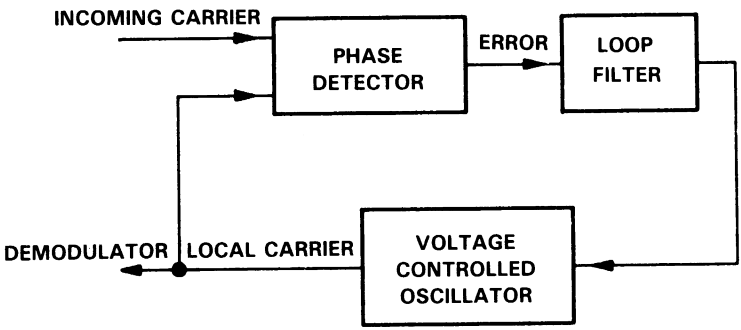


Figure 8. Carrier Recovery Phase-Locked Loop

For a microprocessor implementation, the blocks in Figure 6 are implemented digitally. The digital implementation is discussed in the Carrier Recovery Implementation subsection of "Functions Implemented in the TMS32010." Only the issues associated with the carrier recovery phase-locked loop are considered here.

The phase detector (PD) generates an error signal that is used to synchronize the local carrier to the incoming carrier. This error signal must contain the information about the phase and frequency difference between the local and the incoming carriers. To implement the correct carrier recovery algorithm, it is critical to know the exact dependence of the phase detector output on the frequency and phase difference between the two carriers (discussed later in this subsection). The phase detector output is of the form[7,8]

$$E(nT_b) = \hat{Q}(nT_b)\, I'(nT_b) - \hat{I}(nT_b)\, Q'(nT_b) \tag{8}$$

where $\hat{I}$ and $\hat{Q}$ are the I and Q channel decisions and T_b is the baud period (1/600 s). If the decisions are correct, then

$$\hat{Q}(nT_b) = Q(nT_b) \tag{9}$$

$$\hat{I}(nT_b) = I(nT_b) \tag{10}$$

i.e., the outputs of the transmitter coder (see Figure 3) have been successfully recovered. The probability that these decisions are correct is maximum in the middle of each baud because the incoming signal energy is maximum here. Based on the error signal $E(nT_b)$, the local carrier is corrected once every baud, i.e., at a 600-Hz frequency. Geometrically, the error $E(nT_b)$ is a measure of the geometrical distance between the point used to make the decision and the optimum one. The optimum decision points are the constellation points. It is shown later that when the local carrier has the same phase and frequency with the incoming carrier, the error $E(nT_b) = 0$. In this case, the point used to make the decision coincides with a constellation point. The optimality of the receiver constellation points is discussed next.

Optimality in the receiver, in terms of low probability of error, is determined only by the geometrical distance between the constellation points.[9] The four constellation points of Figure 2, notated as A, B, C, and D, are optimum. The following intuitive argument helps to illustrate this. The four points lie on a circle of normalized unity radius. In the configuration of Figure 2, point A is equidistant from points B and D. This means that the probability of error p when deciding between points A or B, i.e., deciding point A when point B is correct and vice versa, is equal to the probability of error when deciding between points A or D. If point A moves counterclockwise, it moves away from point B but closer to point D. Since at the new location, point A is farther from point B, the probability of error p_1 when deciding between points A or B decreases, i.e., $p_1 < p$. However, at this new location, point A is closer to point D, and therefore, the probability of error p_2 when deciding between points A or D increases, i.e., $p_2 > p$. Using the analytical tools discussed in [9], it can be shown that $p_1 + p_2 > 2p$. Since the overall probability of error increases ($p_1 + p_2 > 2p$) if point A moves away from the location indicated in Figure 2, the resulting structure is no longer optimum. This is not true, however, if all four constellation points are equally rotated by an arbitrary amount in the clockwise or counterclockwise direction. Therefore, an infinite set of constellation points that preserve optimality in the receiver exist. The final choice depends on implementation considerations.

For the modem implementation described in this report, two considerations lead to a 45-degree rotation (see Figure 9) of the transmitter constellation diagram of Figure 2.

1. For the constellation points of Figure 9, the decision boundaries are the I and Q axes. That is, the decision region for point A is the first quadrant, the decision region for point D the second quadrant, and so on. Therefore, a decision can be made based only on the sign of the demodulated I ($I'(nT_s)$) and Q ($Q'(nT_s)$) channels.
2. For this set of constellation points, the products $\hat{Q}(nT_b)\ I'(nT_b)$ and $\hat{I}(nT_b)\ Q'(nT_b)$, required to calculate the phase error $E(nT_b)$ (see equation (8)), obtain on the average maximum values. Therefore, an optimum utilization of the dynamic range is achieved, and the error function calculated by (8) is the least-noise sensitive.

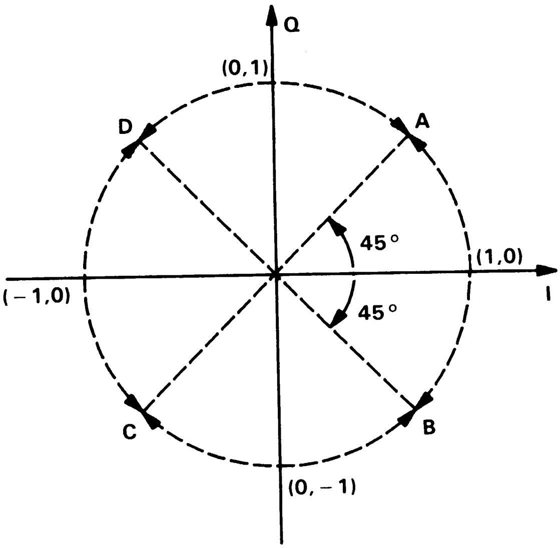


Figure 9. Modem Receiver Decision Points

The error $E(nT_b)$, the output of the phase detector, as given by (8) shows no apparent dependence on the phase or frequency difference between the local and incoming carriers. The discussion that follows shows the dependence of $E(nT_b)$ on the phase difference between the two carriers. This discussion is then extended to include the case of frequency as well as phase difference.

The inputs $I_p(nT_s)$ and $Q_p(nT_s)$ of the receiver demodulator (see Figure 6) are given by (see Appendix A)

$$I_p(nT_s) = I(nT_s)\cos(\omega nT_s + \theta_r) + Q(nT_s)\sin(\omega nT_s + \theta_r) \quad (11)$$

$$Q_p(nT_s) = I(nT_s)\sin(\omega nT_s + \theta_r) - Q(nT_s)\cos(\omega nT_s + \theta_r) \quad (12)$$

where ω and θ_r are the incoming (received) carrier frequency and phase, respectively. The outputs $I'(nT_s)$ and $Q'(nT_s)$ of the demodulator are described by equations (5) and (6), respectively. Introducing an arbitrary phase θ_l in the local carrier, (5) and (6) can be rewritten as

$$I'(nT_s) = I_p(nT_s)\cos(\omega' nT_s + \theta_l) + Q_p(nT_s)\sin(\omega' nT_s + \theta_l) \tag{13}$$

$$Q'(nT_s) = I_p(nT_s)\sin(\omega' nT_s + \theta_l) - Q_p(nT_s)\cos(\omega' nT_s + \theta_l) \tag{14}$$

where ω' is the local carrier frequency.

Assuming no frequency difference ($\omega' = \omega$), substitution of (11) and (12) into (13) and (14) gives

$$I'(nT_b) = I(nT_b)\cos(\theta_e) + Q(nT_b)\sin(\theta_e) \tag{15}$$

$$Q'(nT_b) = Q(nT_b)\cos(\theta_e) - I(nT_b)\sin(\theta_e) \tag{16}$$

where $\theta_e = \theta_r - \theta_l$ is the phase difference between the two carriers. Note that if $\theta_r = \theta_l$, then $\theta_e = 0$. From (15) and (16),

$$I'(nT_b) = I(nT_b)$$

$$Q'(nT_b) = Q(nT_b)$$

i.e., the output of the receiver demodulator at the middle of the baud is the same as the output of the transmitter coder (baseband information).

Assuming the decisions are correct, equations (9) and (10) hold. Substitution of (9), (10), (15), and (16) into the error signal defined by (8) gives

$$E(nT_b) = \{I^2(nT_b) + Q^2(nT_b)\}\sin(\theta_e) \tag{17}$$

The quantity $I^2(nT_b) + Q^2(nT_b)$ is a positive quantity (sum of squares). With

$$I^2(nT_b) + Q^2(nT_b) = K,$$

(17) can be rewritten as

$$E(nT_b) = K\sin(\theta_e) \qquad \text{where } K > 0 \tag{18}$$

Equation (18) is the same as (8) under the assumption of correct decisions ((9) and (10)). However, the phase information is more apparent in (18) than in (8), and leads to the following algorithm for the carrier recovery: If the phase of the received carrier is greater than the phase of the local carrier ($\theta_r > \theta_l$), the phase error θ_e is positive ($\theta_e > 0$). From (18), this implies that the output of the phase detector is also positive ($E(nT_b) > 0$). Therefore, if $E(nT_b) > 0$, the phase of the local carrier must be advanced, resulting in a smaller phase error. On the other hand, if the phase of the received carrier is less than the phase of the local carrier ($\theta_r < \theta_l$), the phase error is negative ($\theta_e < 0$). From (18), this implies that the output of the phase detector is also negative ($E(nT_b) < 0$). Therefore, if $E(nT_b) < 0$, the phase of the local carrier must be delayed.

In the case of frequency as well as phase difference, a similar development leads to

$$E(nT_b) = K \sin(\omega_e nT_b + \theta_e) \qquad \text{where } K > 0 \tag{19}$$

where $\omega_e = \omega - \omega'$ is the frequency difference between the incoming and local carriers. Since this frequency is very small (on the order of a few Hz) and the phase error correction is applied every baud (600 Hz), the term $\omega_e nT_b$ can be considered to be constant and the term $\omega_e nT_b + \theta_e$ in (19) an overall phase error. Therefore, using the algorithm discussed earlier, the frequency difference is compensated for as phase difference. Also note that in (19), $E(nT_b) = 0$ when $\omega_e = 0$ and $\theta_e = 0$; i.e., the local carrier is completely synchronized with the incoming carrier. Therefore, the error signal $E(nT_b)$ generated by the phase detector contains the information about the frequency and phase difference between the incoming and local carriers.

The error signal $E(nT_b)$ generated by the phase detector is processed by the loop filter as shown in Figure 8. Only the DC and low-frequency components of this signal must drive the Voltage Controlled Oscillator (VCO).[6] Therefore, the loop filter is basically a lowpass filter, whose most important characteristic is its bandwidth.

A large bandwidth of the loop filter implies that high-frequency components pass through the filter. Since the high-frequency information is applied to the VCO, the local carrier quickly locks-on to the incoming carrier. However, noise also passes through the filter, and the Bit Error Rate (BER) of the receiver increases. A narrow bandwidth decreases the BER but the lock-on time increases. An intelligent solution consists of starting with a wide bandwidth and, after the receiver is locked-on to the incoming carrier, narrow it down. This approach is used in this implementation and is described further in the Carrier Recovery Implementation subsection of "Functions Implemented in the TMS32010."

Clock Recovery

The purpose of the Clock Recovery block in Figure 5 is to detect the middle of each baud. Once this is known, the decision block can make decisions with minimum probability of error because the energy of the incoming signal is maximum at the middle of the baud. The following paragraphs discuss a robust clock recovery approach.

As the demodulation point moves from one constellation point to another, at least one of the two channels is expected to cross zero (see Figure 9). This zero crossing indicates the beginning of a new baud interval. Therefore, one approach is to look at the zero crossings of the $I'(nT_s)$ and/or $Q'(nT_s)$ channels. However, there are two problems with that approach:

1. From (15) and (16), it can be seen that the presence of a phase difference θ_e between the two carriers can cause severe distortion of the zero crossings. To illustate this point, consider the first of the two equations, repeated here for convenience.

   $$I'(nT_b) = I(nT_b) \cos(\theta_e) + Q(nT_b) \sin(\theta_e)$$

The correct zero crossing information lies in $I(nT_b)$. Multiplication by $\cos(\theta_e)$ scales the $I(nT_b)$ curve, but does not change the location of the zero crossings. This is accomplished by the second additive term $Q(nT_b)\sin(\theta_e)$, which moves the scaled curve up or down depending on the term's sign.

2. The quantization noise in a digital implementation may result in undesirable nonlinearities and mislocation of the zero crossings. This is because finding the zero crossings involves monitoring the change of the sign of a particular variable (I channel and/or Q channel). A zero crossing occurs when this variable changes from a small positive value to a small negative value, and vice versa. Since the quantization noise can seriously affect small quantities (numbers), mislocation of the zero crossings may result.

The first of the above problems indicates that a clock recovery approach is required that is independent of the phase or frequency difference between the two carriers. This becomes clearer by considering the operation of the modem. The first task that the receiver must perform is to adjust the baud clock. During this adjustment, the two carriers are most likely to have a phase and/or frequency difference. Then, once the baud clock is adjusted, the carrier recovery algorithm places the local carrier in phase and in frequency with the incoming carrier.

Consider the energy of the incoming signal

$$\text{Energy} = I'^2(nT_s) + Q'^2(nT_s) \tag{20}$$

Substitution of (15) and (16) into (20), and the use of the identity

$$\sin^2(\theta_e) + \cos^2(\theta_e) = 1$$

gives

$$\text{Energy} = I^2(nT_s) + Q^2(nT_s) \tag{21}$$

This is the energy sent out by the transmitting modem. Equation (21) shows that the energy is independent of any phase and/or frequency difference between the two carriers. Geometrically, the energy is the square of the length of the vector that has its beginning at the intersection of the I and Q axis of Figure 9 and its tip at the demodulation point plotted on the constellation diagram. The path traced by the tip of the energy vector for a series of four consecutive baud intervals, each corresponding to a 90-degree phase change, is shown in Figure 10.

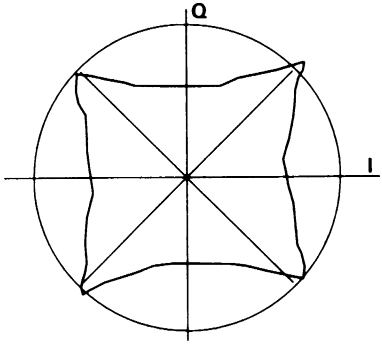


Figure 10. Trace of Demodulation Point Plotted on the Constellation Diagram

The plot shown in Figure 10 was obtained using a simulator. It can be seen that the signal energy ($I^2 + Q^2$) achieves its maximum value at the middle of each baud. Before and after the middle of the baud, the length of this vector is less than maximum. If there is a transition from one quadrant to another, this vector goes through a minimum, thus indicating the beginning of a new baud interval. Only if the same constellation point is transmitted because of a zero-degree phase change does such a transition not occur. It is easy to explain now why a series of zero-degree phase changes can create problems for the receiver. Zero-degree phase changes imply that the transmitter keeps sending the same constellation point. Therefore, the energy vector at the receiver does not go through a minimum for a series of baud intervals; i.e., during these intervals the receiver cannot adjust the baud clock and therefore may lose lock. This situation is avoided with the inclusion of the scrambler in the transmitter structure.

The frequency of the energy minima is discussed next. From Figure 9, it can be seen that there are four possible transitions for each constellation point. For example, consider constellation point A. The four possible transitions are: from point A to B, from A to C, from A to D, and from point A to A (i.e., receiving a zero-degree phase change). Three out of the four possible transitions result in a quadrant change (transitions from point A to points B, C, or D). For these transitions, the energy vector goes through a minimum. The fourth transition (from point A to itself), does not result in a quadrant change, but due to the presence of the scrambler, the probability of its occurrence is less than 0.25 (one out of four). Therefore, the average frequency of these minima is greater than 450 Hz for a 600-Hz baud frequency.

For the baud clock adjustment, the advantages of the energy-based approach over the zero crossings-based approach are:

1. The energy-based approach is independent of the phase and frequency difference between the two carriers, and therefore it gives the correct information about the incoming baud boundaries.

2. The average frequency of the energy minima is greater than 450 Hz while the average frequency of the zero crossings of the I or Q channels is between 300 and 400 Hz. The explanation follows. In the four possible transitions for each constellation point, two of them result in a zero crossing for a particular channel. Considering, for example, the transitions of constellation point A of Figure 9, the transitions from point A to points C or D result in a zero crossing for channel I. The transitions from point A to points B or C result in a zero crossing for channel Q. This implies that for a baud frequency of 600 Hz, the frequency of the zero crossings of a particular channel (I or Q) is on the average 300 Hz (two out of four). Because of the scrambler, the probability of retransmitting the same constellation point (zero-degree phase change) is minimized. This implies that on the average the frequency of the zero crossings of a particular channel increases. In the limit (no zero-degree phase changes), the average frequency of the zero crossings approaches 400 Hz (two out of three). Therefore, the average frequency of the zero crossings of a particular channel is between 300 and 400 Hz. To obtain more information from the zero crossings (greater average zero crossings frequency), the zero crossings of both the I and Q channels must be considered. However, this approach involves monitoring two quantities (I channel and Q channel) compared to monitoring only one (energy) if the energy-based approach is used.
3. Using the energy-based clock recovery technique described in the Baud Clock Alignment Implementation subsection of "Functions Implemented in the TMS32010," the quantization noise effects are less severe compared to those of a zero crossing-based approach.

Modem Hardware Description

A brief description of the hardware used for the implementation of the Bell 212A/V.22 modem is covered in this section. Most of the signal processing required for the implementation of the modem functional blocks described in the previous section is performed digitally by the TMS32010 digital signal processor (see "Functions Implemented in the TMS32010"). The DTE interface and the protocol are handled by the TMS7742[10,11], an 8-bit EPROM microcomputer with an on-chip UART. Therefore, the hardware required for the system is minimal and consists primarily of the TMS32010 and TMS7742 processors, their memory, the A/D and D/A converters, and the associated antialiasing and smoothing filters.

To aid in the development and prototyping of this project, off-the-shelf development tools were used to build the modem hardware. The TMS32010 and the TMS7742 were emulated using Extended Development Systems (XDS) (part # TMDS3262211 for the TMS32010, and part # TMDS7062230 for the TMS7742). For the A/D and D/A conversions, the TMS32010 Analog Interface Board (AIB) (part # RTC/EVM320C/06) was used. The Cermetek CH1810, Data Access Arrangement (DAA) approved by the Federal Communications Commission (FCC), is used for the telephone-line interface. A block diagram of the modem system hardware is shown in Figure 11.

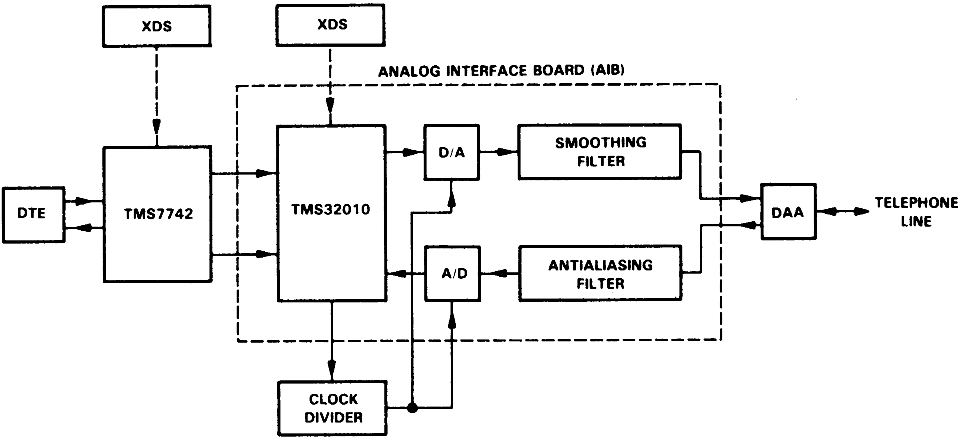


Figure 11. Modem Hardware Block Diagram

Analog Interface

Compensative gain circuits have been placed between the DAA and the analog-to-digital converter. The gain circuit on the receiver side (see Figure 5) was added to match the peak amplitude of the signal from the phone line (−9 dbm or 0.77 V peak-to-peak) with the maximum range of the analog-to-digital converter (10 V peak-to-peak). This allows as much as possible of the A/D's dynamic range to be used without causing saturation. The gain circuit on the transmitter side (see Figure 3) is designed to attenuate the output of the digital-to-analog converter (10 V peak-to-peak) to a level consistent with the phone system signal strength limits (− 12 dbm or 0.55 V peak-to-peak).

Filters

The analog antialiasing and smoothing filters used by the A/D and D/A converters are sixth-order lowpass filters existing on the AIB, implemented using cascaded second-order opamp filter sections. These filters are designed with cutoff frequencies around 4.7 kHz in order to satisfy the Nyquist criterion requirements of the system.

Data Converters

Analog Devices' AD565A and ADC80, monolithic A/D and D/A converters on the AIB, are configured for a ± 10 V full-scale range and are interfaced to I/O port 2 of the TMS32010. The sampling rate for the conversions is determined by a set of presettable counters configured as frequency dividers. These counters are driven by the TMS32010's CLKOUT signal and produce a periodic sampling clock that initiates A/D and D/A conversions. The sampling frequency used is 9.6 kHz.

TMS32010/TMS7742 Interface

The TMS32010 interfaces to the TMS7742 in parallel as shown in Figure 12.

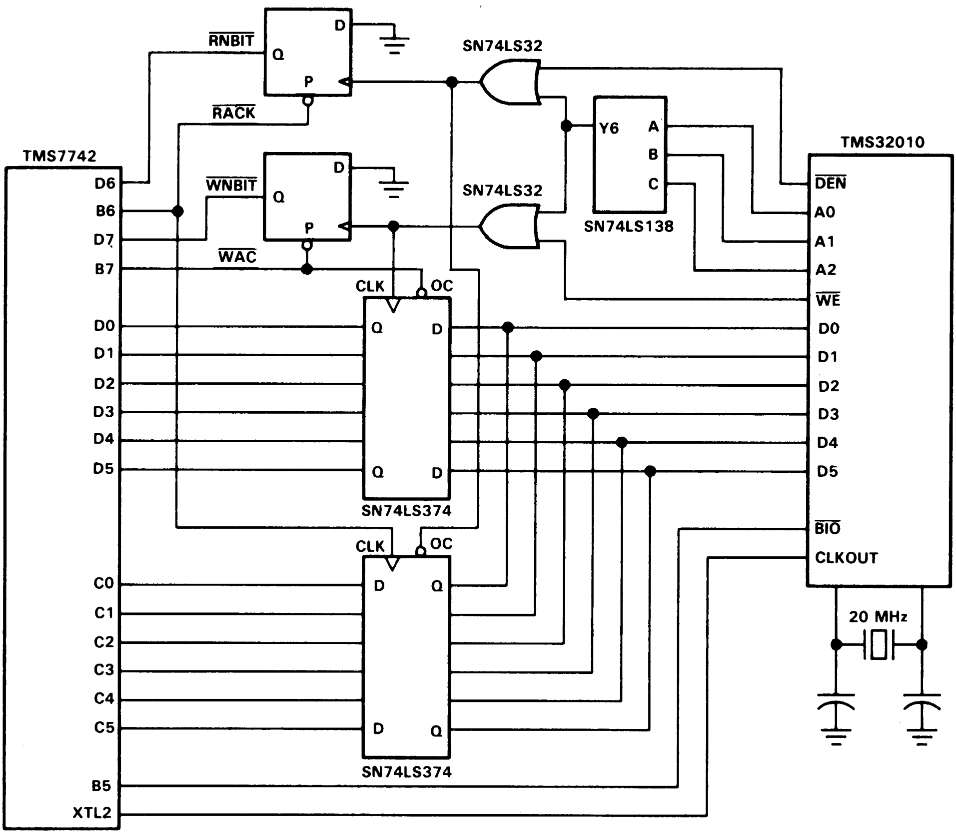


Figure 12. TMS7742 and TMS32010 Interface

The TMS7742 is mapped as an I/O device at Port 6 of the TMS32010. When the TMS32010 writes to Port 6, the $\overline{\text{WNBIT}}$ line goes active (D7). The TMS7742 polls this line and when active, reads data and status bits from the buffer into Port D. It also resets the $\overline{\text{WNBIT}}$ by sending a low pulse to the write acknowledge ($\overline{\text{WACK}}$) line. Similarly, when the TMS32010 reads from Port 6, the $\overline{\text{RNBIT}}$ line goes active. The TMS7742 immediately writes the new data to Port C and resets the read acknowledge ($\overline{\text{RACK}}$) line.

Six bits are used to interface the TMS32010 to the TMS7742. Two of these are used to pass the dibit data, two are used to send commands or status, and the other two are reserved to pass additional data if implementing the V.22 bis. The V.22 bis is a 2400-bps splitband modem that uses quad (four) bits instead of dibits. Table 4 lists the commands. The symbol X is used to indicate a don't care condition.

Table 4. Modem Controller Commands

Bit5	Bit4	Bit3	Bit2	Bit1	Bit0	Command Description
0	0	X	X	X	X	Idle
0	1	X	X	X	X	Run local digital loopback
1	0	X	X	D1	D0	Run modem
1	1	D3	D2	D1	D0	Configure TMS32010 according to D3-D0

In the idle mode, the TMS32010 continues to monitor the commands from the TMS7742. In the local digital loopback mode, the TMS32010 reads the scrambled dibits from the TMS7742 and sends them back to the TMS7742. In the run mode, the TMS32010 reads the scrambled dibits from the TMS7742 and does the required encoding and modulation for the transmission through the telephone network. It also decodes the demodulated data and sends it to the TMS7742 for descrambling. Bits D0 and D1 are used to carry the dibit information. Bits D2 and D3 can be used when implementing the V.22 bis. In the configuration mode, the TMS32010 configures the transmit and receive filters for the originate or answer mode, depending on the data on D3-D0 (see TMS7742 source code provided in Appendix E).

Functions Implemented in the TMS32010

The functions discussed in this section include all of the functional blocks described in the Modem Transmitter and Modem Receiver subsections with the exception of the transmitter scrambler and the receiver descrambler, which are implemented in the TMS7742. Table 5 shows the modem functions that are implemented on each device.

Table 5. Modem Functions Implemented in the TMS32010 and TMS7742

Modem Transmitter	Implemented
Guard Tone Generator	No
Scrambler	TMS7742
Encoder	TMS32010
Digital Lowpass Filters	TMS32010
Originate Mode Modulator	TMS32010
Answer Mode Modulator	No
DTE Interface	TMS7742
Modem Receiver	**Implemented**
Notch Filter	No
Originate Mode Bandpass Filters	TMS32010
Answer Mode Bandpass Filters	No
Automatic Gain Control	TMS32010
Demodulator	TMS32010
Decision Block	TMS32010
Decoder	TMS32010
Descrambler	TMS7742
Clock Recovery	TMS32010
Carrier Recovery	TMS32010
DTE Interface	TMS7742

Each variable used in this section is referred to by its name in the TMS32010 program enclosed in parentheses (see Appendix D).

Transmit Filters

The transmit lowpass filters are implemented using 48-tap FIR structures, whose frequency responses exhibit a raised-cosine shape. A raised-cosine response is a filter response whose pass and stopbands are flat and whose rolloff characteristic is defined as a constant times a (1 + cos) term. The (1 + cos) term results in the rolloff shape being a portion of a cosine wave raised above the X-axis by one, hence the term 'raised-cosine response'. The raised-cosine response is used since it has been shown that it minimizes the intersymbol interference.[12]

The response shape of the transmit filters is actually defined by the square root of a raised-cosine response since the raised-cosine characteristic is split equally between the transmitter and receiver; i.e., both the transmitter and receiver filters are designed to exhibit the square root of the raised-cosine response. This results in the combined, end-to-end response of the path from transmitter to receiver being the full raised-cosine response.

The frequency-response characteristic of the transmit lowpass filters, as shown in Figure 13, rolls off smoothly to approximately -40 dB at 600 Hz.

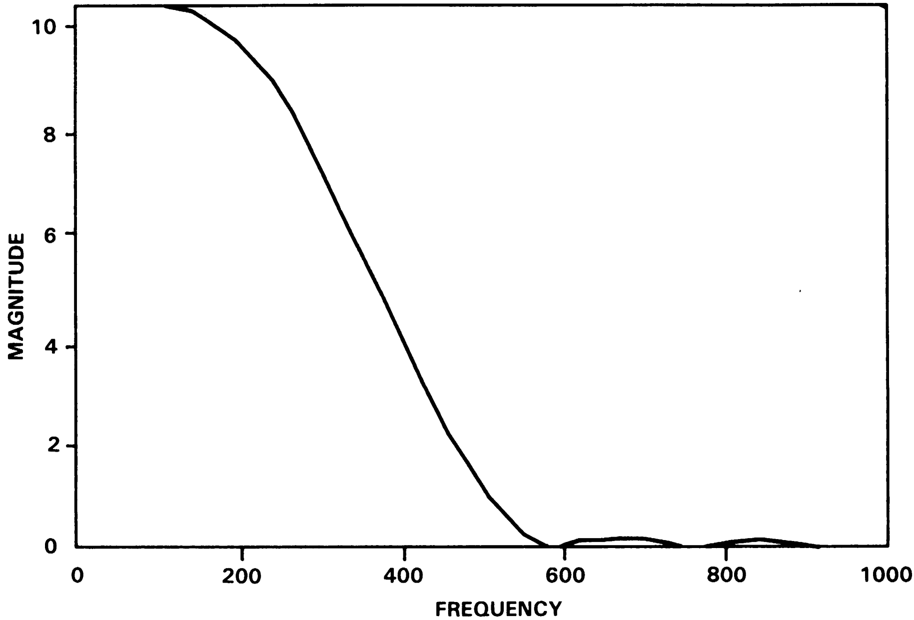


Figure 13. Frequency Response Characteristics of Transmit Lowpass Filters

The FIR structure is well suited to implementation of these filters, because FIR filters are stable, simple in structure, and can be designed to exhibit linear phase. These filters are easily implemented on the TMS32010 since the processor provides special instructions and architectural features that facilitate this type of algorithm. A signal flowgraph of the FIR filter structure is shown in Figure 14.

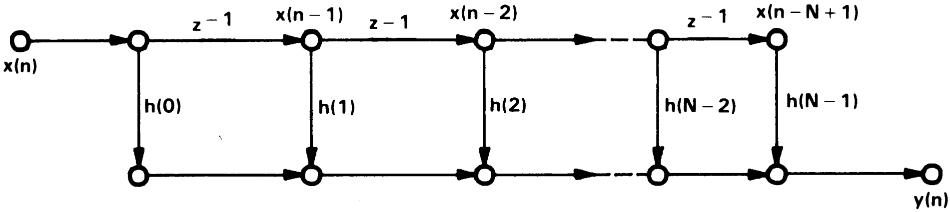


Figure 14. Signal Flowgraph of the FIR Filter Structure

As the flowgraph illustrates, this type of filter uses no feedback, which accounts for its stable behavior. FIR filters implement a transfer function of the form

$$H(z) = B_0 + B_1 z^{-1} + B_2 z^{-2} + B_3 z^{-3} + \ldots + B_n z^{-n} \qquad (22)$$

The parameters in (22) that determine the characteristics of the specific filter implemented are the B coefficients B_0-B_n. In the case of the modem filters, the primary task in designing the filter is the determination of these coefficients so that the filter has the desired response shape, in this case, the raised-cosine response shape. For a detailed description of FIR and IIR filter design for the TMS32010, refer to "Implementation of FIR/IIR Filters with the TMS32010/TMS32020," an application report[13], and to *Digital Filter Design*, a book by T.W. Parks and C.S. Burrus.[14]

The raised-cosine response shape is defined by

$$H(f) = \begin{cases} \dfrac{1}{2\,B_t} & |f| < f_l \\ \dfrac{1}{4\,B_t}\left\{1 + \cos\left[\dfrac{\pi\,(\,|f| - f_l\,)}{2\,B_t - 2\,f_l}\right]\right\} & f_l < |f| < 2\,B_t - f_l \\ 0 & |f| > 2\,B_t - f_l \end{cases} \qquad (23)$$

where $f_l = (1 - p)\,B_t$.

For this design, $B_t = 300$ Hz and $p = 0.75$. Note that (23) describes the ideal zero-phase version of the raised-cosine response.

The actual frequency response of the transmit filters, shown in Figure 13, is the square root of the raised-cosine response described by (16).

To calculate the B coefficients required to implement this response in an FIR filter, the square root of (23) is first calculated. The Inverse Fourier Transform of the response is then used to generate the time-domain representation of the filter transfer function (the impulse response of the filter). In an FIR filter, the impulse response of the filter corresponds directly to the filter coefficients. Therefore, obtaining the coefficients requires merely shifting the impulse response in time to obtain a linear-phase version of the filter, and then sampling the impulse response at the system sampling rate.

After the filter coefficients are obtained, implementation of the filter digitally in the TMS32010 is accomplished by directly translating the signal flowgraph of Figure 14 into assembly language code.

As shown in Figure 14, the output of the filter is defined to be the sum of each of the delayed versions of the input, multiplied by the appropriate coefficient. In the TMS32010, the delayed versions of the previous input samples are stored in a table with the oldest sample stored at the highest address and the newest sample stored at the lowest address.

In the TMS32010 implementation, the transmit filters are arranged in a somewhat different manner from that which is commonly used for digital filters. In many digital filters, the input sample rate is the same as the output sample rate. In the transmit filters, however, the input sample rate is reduced because the rate of change of the information entering the filter is known to be slower than the filter sample rate. Filters of this type are known as interpolating filters, and the ratio of the output sample rate to the input sample rate is referred to as the interpolation factor. In the modem transmit filters, the input sample rate is 600 Hz (the baud rate), and the output sample rate is 9.6 kHz, resulting in an interpolation factor of 16. As a result, the input of the filter is updated only after every 16 output samples, and is zero otherwise. Thus, the effective input $a(nT_S)$ to the transmit filters can be described for the I channel as

$$a(nT_S) = \begin{cases} I\left(\frac{nT_b}{L}\right) & \text{for } n = 0,\ \pm L,\ \pm 2L, \text{ etc.} \\ 0 & \text{otherwise} \end{cases} \tag{24}$$

and for the Q channel as

$$a(nT_S) = \begin{cases} Q\left(\frac{nT_b}{L}\right) & \text{for } n = 0,\ \pm L,\ \pm 2L, \text{ etc.} \\ 0 & \text{otherwise} \end{cases} \tag{25}$$

This technique reduces the number of multiplications required to compute the filter output from N to N/L where N is the length of the filter and L is the interpolation factor.

In both the transmit and receive filters, sampling of the nonzero portion of the filter impulse response at the system sample rate results in only 37 taps required for proper implementation of the filters. However, since the transmit filters are interpolating filters with an interpolation factor of 16, 16 taps are processed for each sample of the input. As a result, the number of taps in the filter must be an integer multiple of 16. In this case, 48 actual taps are used.

With a 48-tap filter and an interpolation factor of 16, only three multiplies are required to calculate the output of the filter. Because of this, these filters are coded on the TMS32010 somewhat differently than FIR filters that are not interpolated. In most filters, the data is shifted each time a sample is processed. With interpolation, the data is shifted only when a new input is processed, i.e., every 16 samples. During the remaining samples (when a new input is not being received), instead of shifting the data, a pointer (XPTR) is shifted through the table of coefficients so that effectively the coefficients are shifted. Thus, the complete filter output can be calculated with the following short section of code:

```
ZAC                         * CLEAR ACCUMULATOR
LT      XIBUF2              * LOAD OLDEST SAMPLE
MPY     CX2                 * MPY BY COEFF 2
LTD     XIBUF1              * LOAD NEXT SAMPLE
MPY     CX1                 * MPY BY COEFF 1
LTD     XIBUF0              * LOAD NEWEST SAMPLE
MPY     CX0                 * MPY BY COEFF 0
APAC                        * MAKE FINAL SUM
SACH    XIOUT,1             * STORE OUTPUT
```

This code calculates the output of the I channel filter when a new input sample is being processed. The code that implements the filter output calculation when a new sample is not being input is similar to this code except that LTA instructions are used in place of LTD instructions.

During samples in which new inputs are being received, the inputs and the coefficients are shifted. This results in savings in data RAM space since only three data values must be stored.

Receive Filters

The receiver bandpass filters are implemented using 37-tap FIR structures, which also exhibit a raised-cosine frequency response characteristic. These filters are virtually identical in structure to the transmit lowpass filters, with the exceptions that the cutoff frequencies are different and the receive bandpass filters do not interpolate since the input and ouput sample rates are the same. Like the transmit lowpass filters, the actual response implemented in these filters is the square root of the raised-cosine response since this response is split equally between the transmit and receive sections. The receive filters are centered around the carrier frequency f_C (1200 Hz for originate and 2400 Hz for answer), and roll off smoothly to approximately -40 dB at $f_C \pm 600$ Hz. The frequency response characteristic of these filters is shown in Figure 15.

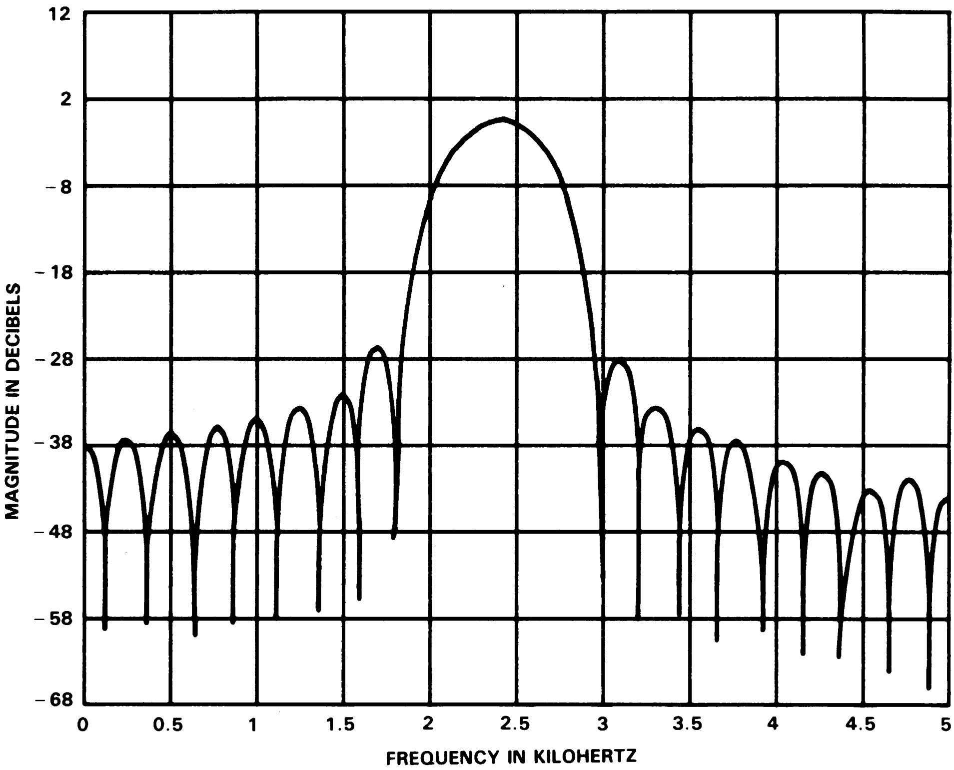


Figure 15. Frequency Response of Receiver Bandpass Filters

Except for the difference in filter order (the number of taps in the filter), the signal flowgraph and transfer function equation for the receive filters are identical to those of the transmit lowpass filters shown in Figure 13 and described by equation (22), respectively.

Besides being similar in structure to the transmit lowpass filters, the receive filters are actually designed directly from the transmit filters by simply shifting the filters' center frequencies. This is possible because the bandwidth of the transmit filters is the same as that required for the receive filters, and the transmit filters exhibit the raised-cosine response also required for the receive filters.

In order to generate the proper coefficients to implement the receive filters, the coefficients of the transmit filter are multiplied by a sine wave to obtain the I channel coefficients and by a cosine wave to obtain the Q channel coefficients. Specifically, if $h(nT_s)$ are the transmit filter coefficients, the receive filter coefficients $h_1(nT_s)$ and $h_2(nT_s)$ are obtained by

$$h_1(nT_S) = 2\, h(nT_S) \cos(n\omega T_S) \qquad \text{(I channel)}$$

$$h_2(nT_S) = 2\, h(nT_S) \sin(n\omega T_S) \qquad \text{(Q channel)} \qquad (26)$$

where T_s is the sampling period. Note that the factor of two must be included for the original frequency spectrum to be translated to the new frequency with the same magnitude.

The result of multiplying the transmit filter coefficients by sine and cosine is to effectively modulate their frequency response characteristics by a carrier at the frequency of the sine and cosine waves. This translates the frequency spectrum of the resultant filter up in frequency to a point centered around the frequency of the modulating signal, which is precisely what is required for the receive bandpass filters. Accordingly, bandpass filters for the originate mode are multiplied by sine and cosine functions at 1200 Hz and those for the answer mode are multiplied by sine and cosine functions at 2400 Hz, thus yielding the exact filters required.

In addition to shifting the frequency spectrum of the filters to the appropriate center frequencies, the fact that the I channel filter is multiplied by a sine function and the Q by a cosine function results in another important characteristic of these filters; the outputs of these filters are exactly 90 degrees out of phase with respect to each other. This provides a convenient method for implementing the phase shift required for proper demodulation of the I and Q channels. Also, since the filters are symmetric FIR structures, their phase response is linear, and the difference in phase shift between the two filters is precisely 90 degrees. This is beneficial because deviations from a precise 90-degree phase shift can cause serious distortion in other parts of the modem receiver.

In a direct implementation of this type of filter on the TMS32010, the filter output is calculated by repeatedly using the following two-instruction sequence:

```
LTD          * LOAD T, ACCUMULATE, DATA SHIFT
MPY          * MULTIPLY BY NEW COEFFICIENT
```

This sequence performs the following four operations:

1. Loads the T register with the input value,
2. Multiplies the input value by the appropriate coefficient,
3. Adds the product to the accumulator, and
4. Shifts the input data one place in the table, making room for the next input sample.

For FIR filters, a sequence of pairs of the LTD and MPY instructions is all that is required to implement the complete filter.

In the TMS32010, the receive filters are implemented in a somewhat more conventional manner than the transmit filters. The receive filters do not interpolate; however, due to careful choice of sample points on the impulse response, every second coefficient in each filter is zero, reducing by a factor of two the number of LTD/MPY instruction pairs that must be executed to calculate the filter output.

Another feature of the FIR structure, which simplifies the implementation of these filters, is that since there is no feedback, the delay path ($x(n-1)$, $x(n-2)$,..., in Figure 14) for the two filters contains the same values for input samples in data RAM. Because of this, the same delay path can be used for both the I and the Q channel filters. This reduces by a factor of two the RAM required for data storage. As a result, the code that implements the I channel filter (processed first) uses LTA instructions instead of LTDs, performing no shift of the input data table within memory. A one-position shift of the data table is then performed when the Q channel filter output is calculated.

Even though every other coefficient is zero, each sample in the delay table must still be shifted by one memory location during each pass through the filter. Since the Q channel filter performs this shifting but only operates on every second data point, an additional DMOV instruction is coded between each LTD/MPY instruction pair in order to shift the even-numbered data table entries.

The assembly code that implements the Q channel bandpass filter is shown below.

```
*
*    FIRST (Nth) TAP SETS UP FOR REST OF FILTER
*
        ZAC                         * CLEAR ACCUMULATOR
        LT          RBUF35          * LOAD T REGISTER
        DMOV        RBUF35          * SHIFT OLD VALUE
        MPYK        QCF35           * MPY BY COEFFICIENT
        DMOV        RBUF34          * PERFORM EXTRA SHIFT
*
*    SECOND TAP
*
        LTD         RBUF33          * LOAD T, ACCUMULATE, DATA SHIFT
        MPYK        QCF33           * MULTIPLY BY COEFFICIENT
        DMOV        RBUF32          * PERFORM EXTRA SHIFT
          .
          .
          .
*
*    LAST TAP
*
        LTD         RBUF1           * LOAD T, ACCUMULATE, DATA SHIFT
        MPYK        QCF1            * MULTIPLY BY COEFFICIENT
        DMOV        RBUF0           * PERFORM EXTRA SHIFT
*
        APAC                        * ADD LAST SUM
        SACH        QSUM,4          * STORE FILTER OUTPUT
```

Automatic Gain Control Implementation

To better control the signal strength of the receiver, a software Automatic Gain Control (AGC) algorithm was added. The need of an AGC stems from the use of thresholds in both the carrier recovery and clock recovery algorithms. For increased performance, these thresholds (discussed in the following two subsections) must remain valid (unchanged) for different levels of the incoming signal. This is achieved with the use of the software AGC.

The arrangement of the AGC with respect to the other functional blocks of the modem receiver was shown in Figure 5. The AGC monitors the I channel of the receiver and calculates a gain correction factor. Both the I and Q channels are then multiplied by this gain correction factor so that the signal maxima remain within a certain range. This range is narrow compared to the range of the incoming signal maxima. The peak-to-peak voltage of the incoming signal is between 2 mV and 700 mV. In 16-bit hexadecimal Q15 format,[15] this range is from >5C to >5999. However, with the use of the software AGC, the signal maxima are in the range 780 mV (>6400) to 820 mV (>6900).

The gain correction factor is calculated once every three bauds by a two-step process. First, the three maximum values of the signal (BSMAX), each one corresponding to one baud (16 samples), are monitored and added to each other. A counter (AGCNT) is used to keep the count of the signal maxima. The previous running average is then added to this sum, and the result is divided by four to obtain the new running average (AGCRA). The division by four is accomplished by shifting the final sum, contained in the accumulator, two locations to the right before storing it in the memory as the new running average (AGCRA). The section of code that implements this step is listed below.

```
*
*       DETECT MAX SIGNAL STRENGTH OF I CHANNEL PER BAUD
*                 (THIS CODE IS EXECUTED EVERY CYCLE)
*
AGCAL     EQU    $
          LAC    ISUM       * AGC VALUE CALCULATED USING ISUM
          ABS               * GET MAGNITUDE OF SIGNAL
          SUB    BSMAX      * COMPARE TO PREVIOUS MAX VALUE
          BLZ    OVRMAX     * IF LESS THAN, THEN SKIP UPDATE
          ADD    BSMAX      * RESTORE VALUE AND
          SACL   BSMAX      * STORE AS NEW MAX
*
*        MULTIPLY I  AND Q  CHANNELS BY AGC FACTOR
*
OVRMAX
               .
               .
               .
```

```
*
*       UPDATE THE RUNNING AVERAGE ONCE EVERY THREE BAUDS
*              (THIS CODE IS EXECUTED ONCE EVERY BAUD)
*
AGCUPT    ZALH   AGCRA      * ADD THE NEW BSMAX VALUE
          ADD    BSMAX,14   * TO THE RUNNING AVERAGE
          SACH   AGCRA      * AND SAVE IT
          LAC    AGCNT      * DECREMENT RUNNING AVERAGE
          SUB    ONE        * SAVE IT AND
          SACL   AGCNT      * CHECK FOR ZERO
          SACH   BSMAX      * ZERO OUT RUNNING SIGNAL MAX
          BZ     OVROUT     * IF ZERO, THEN UPDATE AGC
          RET               * ELSE RETURN TO CALLING SEQUENCE
OVROUT    LACK   3          * RESET RUNNING AVERAGE COUNT
          SACL   AGCNT      * TO THREE
          LAC    AGCRA      * MOVE AGCRA
          SACL   AGCLEV     * TO THE CALCULATION LEVEL
          LAC    AGCRA,14   * DIVIDE RUNNING AVERAGE SUM
          SACH   AGCRA      * BY 4 TO GET NEW RUNNING AVERAGE
```

At the second step, the gain correction factor (AGC) is calculated, based on the running average. A brute force approach is to divide the maximum-allowed signal level by the running average and obtain the gain correction factor as the result of this division. The maximum value of the product of the signal times the gain correction factor should then remain close to the maximum-allowed signal level. However, since divisions are costly in processing time, the second step is implemented by using the running average as an index (AGCLEV) to a 32-word lookup table. The offset to this table (AGCOFF) is added to the index (AGCLEV) to calculate the table entry on which the gain correction factor (AGC) is located. The TBLR instruction is then used to transfer the gain correction factor from program memory to data memory. To lessen the code space required to handle the AGC lookup table, the code uses only the six most significant bits of the running average. This requires a 64-word lookup table. However, since the most significant bit of the six bits is always one, only 32 entries of the table are needed. The gain correction factor, obtained by the table lookup, is shifted so that the product of the gain correction factor times the incoming signal is in Q14 format (designer's choice). The shift factor is provided by the BASIC program used to generate the AGC lookup table (see Appendix C). The TMS32010 code that implements the calculation of the gain correction factor is shown below.

```
          LAC    AGCLEV     * GET AVERAGE MAX SIGNAL LEVEL
          SUB    ONE,14     * COMPARE TO 16384
          BLZ    ASHF1      * IF LESS THAN SHIFT TABLE LOOKUP
          LAC    AGCLEV,7   * GET LOOKUP VALUE
          SACH   TEMP       * MOVE LOOKUP VALUE TO
```

```
        LAC    TEMP        * THE LOW HALF OF THE ACC
        ADD    AGCOFF      * ADD IN TABLE OFFSET
        TBLR   AGC         * AND GET AGC VALUE
        LAC    AGC,15      * DIVIDE THE AGC VALUE
        SACH   AGC         * BY 2 TO FORCE TO Q14 MODE
        RET                * RETURN TO CALLING SEQUENCE
ASHF1   ADD    ONE,13      * COMPARE TO 8192
        BLZ    ASHF2       * IF LESS THAN SHIFT TABLE LOOKUP
        LAC    AGCLEV,8    * GET LOOKUP VALUE
        SACH   TEMP        * MOVE LOOKUP VALUE TO
        LAC    TEMP        * THE LOW HALF OF THE ACC
        ADD    AGCOFF      * ADD IN TABLE OFFSET
        TBLR   AGC         * AND GET AGC VALUE
        RET                * RETURN TO CALLING SEQUENCE
ASHF2   ADD    ONE,12      * COMPARE TO 4096
        BLZ    ASHF3       * IF LESS THAN SHIFT TABLE LOOKUP
        LAC    AGCLEV,9    * GET LOOKUP VALUE
        SACH   TEMP        * MOVE LOOKUP VALUE TO
        LAC    TEMP        * THE LOW HALF OF THE ACC
        ADD    AGCOFF      * ADD IN TABLE OFFSET
        TBLR   AGC         * AND GET AGC VALUE
        LAC    AGC,1       * AGC VALUE*2 TO ADJUST
        SACL   AGC         * FOR LOWER SIGNAL STRENGTH
        RET                * RETURN TO CALLING SEQUENCE
          .
          .
          .
ASHF6   ADD    ONE,5       * COMPARE TO 32
        BLZ    NOEDT       * LOST MINIMUM ENERGY LEVEL
        LAC    AGCLEV,13   * GET LOOKUP VALUE
        SACH   TEMP        * MOVE LOOKUP VALUE TO
        LAC    TEMP        * THE LOW HALF OF THE ACC
        ADD    AGCOFF      * ADD IN TABLE OFFSET
        TBLR   AGC         * AND GET AGC VALUE
        LAC    AGC,5       * AGC VALUE*32 TO ADJUST
        SACL   AGC         * FOR LOWER SIGNAL STRENGTH
        RET                * RETURN TO CALLING SEQUENCE
```

The AGC table was generated by the BASIC program listed in Appendix C. This program is written to execute on any MS-DOS operating system. The program prompts the user for the table size and gain range factor, and then generates and stores the AGC table. The table is stored in a format that allows insertion directly into the user's code.

Carrier Recovery Implementation

The carrier recovery is implemented with a phase-locked loop, as explained in the Modem Receiver subsection. In Figure 8, the functional blocks that must be digitally implemented are the phase detector, loop filter, and Voltage Controlled Oscillator (VCO).

Phase Detector

In the middle of each baud, the phase detector block calculates an equation equivalent to (8), repeated below for convenience,

$$E(nT_b) = \hat{Q}(nT_b)\, I'(nT_b) - \hat{I}(nT_b)\, Q'(nT_b)$$

where I′ (RECI) and Q′ (RECQ) are the baseband (demodulated) I and Q channels, and $\hat{I}$ and $\hat{Q}$ are the I and Q channel decisions. The derivation of the equivalent equation to (8) is discussed next.

In Figure 9, the I channel decision for constellation point A is the length of the projection of the vector $\overrightarrow{OA}$ on the I axis. Similarly, the Q channel decision for constellation point A is the length of the projection of the vector $\overrightarrow{OA}$ on the Q axis. Since the four constellation points A, B, C, and D are located on the 45 and −45 degree lines, the lengths of these projections are the same. With this common length denoted by L, the I channel decisions can be expressed as

$$\hat{I}(nT_b) = \begin{cases} +L \text{ for points A and B} \\ -L \text{ for points C and D} \end{cases} \tag{27}$$

The value of L depends on the radius of the circle on which the four constellation points are located. Equation (27) can equivalently be expressed as

$$\hat{I}(nT_b) = \mathrm{sgn}(I'nT_b))\; L \tag{28}$$

where sgn is the sign function defined as

$$\mathrm{sgn}(I'(nT_b)) = \begin{cases} +1 & \text{if } I'(nT_b) > 0 \text{ (points A and B)} \\ -1 & \text{if } I'(nT_b) < 0 \text{ (points C and D)} \end{cases} \tag{29}$$

Similarly,

$$\hat{Q}(nT_b) = \mathrm{sgn}(Q'(nT_b))\; L \tag{30}$$

Substitution of (28) and (30) into (8) gives

$$E(nT_b) = L\, \{\mathrm{sgn}(Q'(nT_b))\, I'(nT_b) - \mathrm{sgn}(I'(nT_b))\, Q'(nT_b)\} \tag{31}$$

Equations (31) and (8) are identical. However, (31) is the final step towards the equation implemented in the TMS32010. Since L in (31) is a positive constant, an equivalent error function that contains the phase and frequency information is

$$E'(nT_b) = \mathrm{sgn}(Q'(nT_b))\, I'(nT_b) - \mathrm{sgn}(I'(nT_b))\, Q'(nT_b) \tag{32}$$

Equation (32) is the one implemented in the TMS32010 as part of the carrier recovery algorithm. In this equation, sgn(I') (SIGNI) and sgn(Q') (SIGNQ) are the I and Q channel decisions, respectively. The TMS32010 code used to implement (32) is shown below.

```
*
*  COMPUTE CARRIER ERROR SIGNAL
*
*  e(t) = RECI*SIGNQ - RECQ*SIGNI
*
COMERR LT      RECI        * T=RECI
       MPY     SIGNQ       * P=RECI*SIGNQ
       LTP     RECQ        * T=RECQ, ACC=RECI*SGNQ
       MPY     SIGNI       * P=RECQ*SIGNI
       SPAC                * ACC=RECI*SIGNQ-RECQ*SIGNI
       SACH    ERROR,1     * STORE IN ERROR
```

Loop Filter

The error signal $E'(nT_b)$ (ERROR), generated by the phase detector (equation (32)), is filtered by the carrier recovery loop filter (see Figure 8). The filter was implemented as a first-order Infinite Impulse Response structure. In other words, the loop filter is just an integrator with transfer function

$$H_1(z) = \frac{B_1}{1 - A_1 z^{-1}} \tag{33}$$

where A_1 (PLL1) and B_1 (PLL2) are the filter coefficients.

A higher-order filter was not used, because high-order filter structures usually introduce more phase delay than first-order sections. Phase delays[16] are critical in the operation of a phase-locked loop, and their effects are difficult to analyze.

The time-domain equivalent of (33) is

$$y(n) = B_1 x(n) + A_1 y(n-1) \tag{34}$$

where x(n) is the input to the filter and y(n) the output. The signal flowgraph of the carrier recovery loop filter is shown in Figure 16.

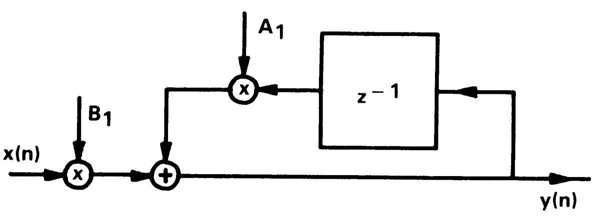


Figure 16. Carrier Recovery Loop Filter

The TMS32010, with a hardware on-chip multiplier, is most efficient in the implementation of such filter structures.[13] The code used to implement the carrier recovery loop filter (equation (34)) is shown below. The filter's input x(n) is stored in ERROR, and the filter's output y(n) is stored in ERRSIG.

```
*
*  LOOP FILTER
*
        LT      PLL2          * T=PLL2
        MPY     ERROR         * P=PLL2*ERROR
        LTP     PLL1          * ACC=PLL2 ERROR, T=PLL1
        MPY     ERRSIG        * P=PLL1*ERRSIG
        APAC                  * ACC=PLL2*ERROR+PLL1*ERRSIG
        SACH    ERRSIG,1      * STORE IN ERRSIG
```

The effect of the loop filter's bandwidth in the modem performance is considered in the following discussion where the bandwidth of the loop filter is defined as the frequency at which the magnitude of the filter's transfer function is 3 db below its maximum value. Therefore, the bandwidth of the loop filter is the frequency ω_b at which

$$20\log \frac{|H_1|\max}{|H_1(\omega_b)|} = 3 \tag{35}$$

where $|H_1|\max$ is the maximum value of the magnitude of the filter's transfer function. Substituting $z = e^{j\omega}$ in (33) gives

$$|H_1(\omega)| = \frac{|B_1|}{\{1 + A_1^2 - 2A_1 \cos(\omega)\}^{1/2}} \tag{36}$$

Equation (36) is maximum when the denominator is minimum. This is true for $\omega=0$, i.e., at DC. Substituting $\omega=0$ in (36) gives

$$|H_1|\max = \frac{|B_1|}{1-A_1} \qquad \text{where } 0 < A_1 < 1 \tag{37}$$

Substitution of (36) and (37) into (35) gives the following quadratic equation that relates the bandwidth of the loop filter ω_b to the coefficient A_1.

$$A_1^2 + 2A_1 \{\cos(\omega_b) - 2\} + 1 = 0 \tag{38}$$

Therefore, the value of the coefficient A_1 determines the bandwidth of the loop filter. Figure 17 shows a plot of the values of A_1 versus the bandwidth ω_b, i.e., a plot of (38).

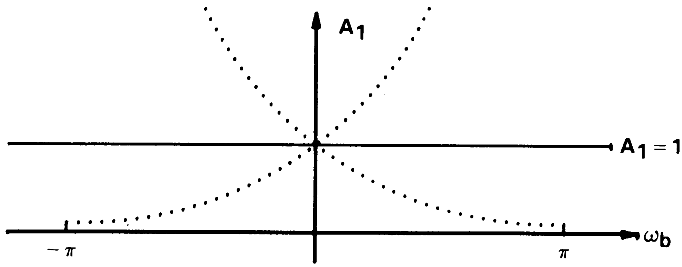


Figure 17. Parameter A_1 vs. the Bandwidth of $H_1(z) = \dfrac{B_1}{1 - A_1 \ z^{-1}}$

The two curves in Figure 17 represent the solutions of the quadratic equation (38). From this figure, it can be seen that the closer A_1 is to unity, the narrower the bandwidth of the filter. If $A_1 = 1$, the magnitude response begins rolling off at zero frequency ($\omega = 0$). However, this situation must be avoided since $A_1 = 1$ results in placing a pole on the unit circle in the z-domain, thereby causing the filter to oscillate. Since only values less than unity of the coefficient A_1 result in a stable filter structure, Q15 format[15] was used to represent A_1.

The bandwidth ω_b is expressed in radians. Since the sampling frequency f_b corresponds to 2π radians, the bandwidth of the loop filter in Hz is given by

$$\text{Bandwidth} = \frac{\omega_b \, f_b}{2\pi} \quad \text{Hz} \tag{39}$$

Since the loop filter runs once every baud, the sampling frequency f_b is

$$f_b = \frac{1}{T_b} = 600 \text{ Hz}$$

with T_b as the baud interval. This frequency should not be confused with the A/D and D/A sampling frequency designated by f_s and having the value of 9600 Hz.

The bandwidth of the loop filter affects the Bit Error Rate (BER) and the time it takes for the modem receiver to lock-on to the incoming carrier. Initially, a large bandwidth results in a fast lock-on while a narrow bandwidth provides a good BER. Therefore, the ability to switch from a large bandwidth to a narrow one results in a better modem design. With the TMS32010, this is easily implemented using the TBLR instruction that transfers data from program memory to data memory.[15] On startup, A_1 (PLL1) is 0.539 or >4500 in Q15 format. This corresponds to a bandwidth of approximately 63 Hz. Once locked-on with the use of the TBLR instruction, the value of A_1 (PLL1) is changed to 0.953 or >7A00 in Q15 format. This corresponds to a bandwidth of approximately 6 Hz. Lock-on criterion is based on the magnitude of the error function calculated by (32) being less than a certain threshold. The need and calculation of this threshold is covered later in this subsection. The TMS32010 code used to switch the loop filter's bandwidth is given below. The fifth bit of RECST is used as a flag, which if set indicates that the local carrier is locked-on to the incoming carrier.

```
        LAC     ONE,4        * CHECK IF LOCAL CARRIER
        AND     RECST        * IS LOCKED. IF SO, SWITCH
        BNZ     CARLCK       * PLL FILTERS' BANDWIDTH
        B       NORMAL       * EXECUTE NORMAL SEQUENCE
*
CARLCK  LACK    PLLC         * CHANGE CARRIER PLL COEF. 1
        TBLR    PLL1
```

Voltage-Controlled Oscillator

Both the carrier used in the transmitter to modulate the data and the one used in the receiver for the demodulation (local carrier) were implemented in the TMS32010 using a 128-point sine table and a routine to drive it.[17] The voltage-controlled oscillator in the phase-locked loop for the carrier recovery generates the local carrier using this 128-point sine table. The frequency of this digital sine wave is 2400 Hz for an originate modem and 1200 Hz for an answer modem.

Carrier Recovery Threshold

The lowpass-filtered value of the error signal generated by the phase detector contains the information about the phase and frequency difference between the local and incoming carriers. If this value (ERRSIG) is positive, the local carrier must be advanced in phase. If negative, the local carrier must be delayed (see the Modem Receiver subsection). Since there are 128 points in the sine table, there is a 360/128 or 2.8125-degree jump going from one table entry to the next. This implies that corrections should not be made unless the magnitude of the error signal is greater than one table entry because redundant corrections introduce inaccuracies and noise. Therefore, the value of this threshold should correspond to the magnitude of the error signal when there is a 2.8125-degree phase error.

An estimate of the threshold can be obtained as described below. The relation of the phase error signal $E(nT_b)$ to the phase error θ_e is given by (18). Substituting 2.8125 for θ_e in (18) and taking the magnitude of both sides gives

$$|E(nT_b)| = |K \sin(2.8125)| \qquad (40)$$

$K\,(I^2+Q^2)$ is the signal energy, i.e., the maximum value of the I and Q channels. This value is set by the Automatic Gain Control. Since the software AGC used in this implementation of the Bell 212A/V.22 limits the signal maxima between 0.78 and 0.82 (see Automatic Gain Control Implementation in the Modem Receiver subsection), K is between 0.78 and 0.82. Using the average value of 0.80 for K, (40) gives

$$|E(nT_b)| = 0.039.$$

The threshold level should be at 0.039 if the gain of the loop filter given by (33) is unity. For DC, the gain G1 of the loop filter is given by (37), repeated below for convenience.

$$G_1 = |H_1|\max = \frac{B_1}{1 - A_1} \qquad \text{where } 0 < A_1 < 1$$

The coefficient B_1 (PLL2) was chosen to be 0.0039 (or >50 in the Q15 format). As explained earlier, once the receiver is locked, the value of coefficient A_1 (PLL1) is 0.953. From (37), the gain G_1 of the loop filter is $G_1 = 0.082$. Therefore, the threshold is scaled down to

Effective Threshold $= 0.039 \times 0.082 = 0.0032$.

This corresponds to >D in Q12, the format used for the threshold (designer's choice). After this initial estimate of the threshold was obtained, the final value of the carrier recovery threshold (TRSHD1), >7, was determined by trial and error. The calculated threshold is greater than the one obtained by trial and error, because of the use of the maximum value of the loop filter's gain in the threshold calculation.

To improve the lock-on characteristics of the modem, a two-level correction was used for the carrier recovery. If the magnitude of the error (ERRSIG) is less than the threshold (TRSHD1), no correction is applied. If the magnitude of the error is greater than the threshold but less than twice the threshold, one sine-table entry correction is applied by incrementing or decrementing the table entry pointer (RALPHA) by one. If the magnitude of the error is greater than twice the threshold value, then a two-table entry correction is applied by incrementing or decrementing the table entry pointer (RALPHA) by two. All of the corrections are applied to advance or delay the local carrier according to the algorithm described in the Modem Receiver subsection.

Baud Clock Alignment Implementation

The purpose of the clock recovery is to identify the baud boundaries and inform the decision block when the middle of each baud occurs and therefore the optimum time to make an error-free decision (see Figure 5). As explained in the Modem Receiver subsection, one approach for clock recovery (adjustment of the baud clock) is to use the energy of the incoming signal. The energy is the sum of the squares of the demodulated I and Q channels (see equation (20)). As implied by (21), this quantity is independent of any phase and/or frequency difference between the incoming and local carriers.

The minima of the signal energy indicate the beginning of a new baud. This can be seen in Figure 18 where the signal energy is plotted every sample for several consecutive baud intervals.

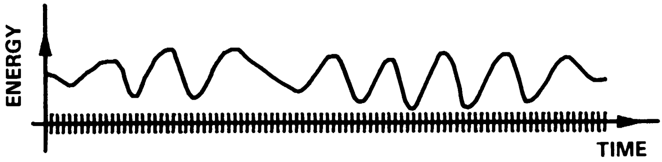


Figure 18. Signal Energy Plotted Every Sample For Several Baud Intervals

Each of the short vertical lines along the horizontal axis in Figure 18 corresponds to a sample time. This data was obtained using the XDS/22 emulator for the TMS32010. The block diagram for the clock recovery algorithm is shown in Figure 19. The functional blocks to be implemented are the error signal generator, loop filter, and baud clock.

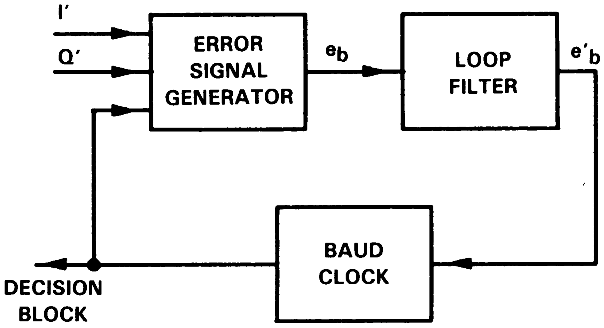


Figure 19. Baud Clock Alignment Block Diagram

Error Signal Generator

The error signal generator calculates the signal energy and from it generates an error signal e_b. This error signal contains the information about how close the local baud boundaries are to the incoming baud boundariers. The error signal is then lowpass-filtered so that noise and high-frequency components are removed. The output of the loop filter corrects the local baud clock.

The critical issue is how to calculate this error signal. Figure 20 shows the signal energy for a single baud interval. This figure was motivated from the realtime data of Figure 18. The 16 energy samples for this baud are indicated as E(0),E(1),...,E(15).

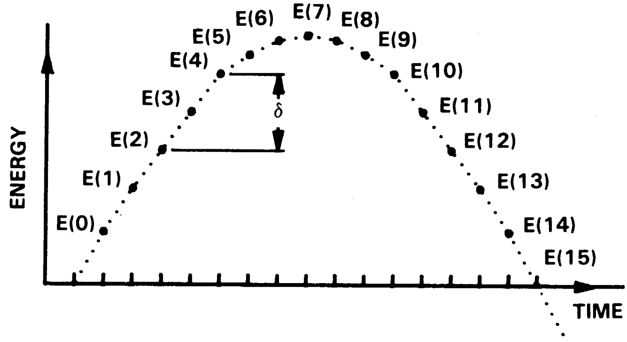


Figure 20. Signal Energy Samples over a Baud

From Figure 20, it can be seen that the energy sample E(7) is located at the middle of the baud (top of the 'energy hill'), and the rest of the samples are located symmetrically around it, i.e., E(6) = E(8), E(5) = E(9), and so on. Therefore, E(7) is taken to be the middle of the local baud. Consider now the difference between the energy sample E(11) that is four samples after E(7) and the energy sample E(3) that is four samples before E(7). If the local baud clock is correctly aligned so that E(7) corresponds to the middle of the incoming baud, then

$$E(11) - E(3) = 0.$$

If

$$E(11) - E(3) > 0,$$

then the sample E(7) is located to the left of the middle of the baud. This means that the middle of the local baud occurred earlier than the middle of the incoming baud. Therefore, the local baud clock must be delayed. On the other hand, if

$$E(11) - E(3) < 0,$$

the middle of the local baud occurred later than the middle of the incoming baud. Therefore, the local baud clock must be advanced.

In summary, the error signal generator computes the signal energy at sample points 3 (PENRGY) and 11 (ENRGY). The sample count information (SAMPLE) is provided by the baud clock shown in Figure 19. The error signal generator then calculates the error signal e_b (BERROR) defined by

$$e_b = E(11) - E(3) \tag{41}$$

The subscript b represents baud since this signal is calculated once every baud.

Loop Filter

Perturbations that may occur in the communications medium pass onto the demodulated I and Q channels. This can be seen from the data of Figure 18 where even with the presence of the automatic gain control, the energy levels are not exactly the same for every baud. Also, the duration of each baud in Figure 18 is not exactly sixteen samples (sixteen short vertical lines) as it theoretically should be. These perturbations result in abrupt changes of the signal generated by the error signal generator. Therefore, the error signal is not directly fed into the baud clock. Instead, it is lowpass-filtered by the loop filter. This removes noise and high-frequency components and results in a stable clock recovery.

The loop filter was implemented as a first-order recursive filter. The transfer function is of the same form as (33).

$$H_2(z) = \frac{B_2}{1 - A_2 z^{-1}} \tag{42}$$

Just as with the loop filter used for the carrier recovery, the most important characteristic of the loop filter used for the clock recovery is its bandwidth. A wide bandwidth results in a quick adjustment of the local baud boundaries to the incoming baud boundaries. A narrow bandwidth results in a more stable clock recovery. A good approach for this filter's design is to start with a wide bandwidth and then switch to a narrow one. All of the information provided in the Carrier Recovery Implementation subsection relating the coefficient A_1 to the loop filter's bandwidth apply here as well. With the use of the TBLR instruction, after the receiver is locked-on to the incoming carrier, the initial wide bandwidth is switched to a narrow one. The initial value of A_2 is 0.5, which is >4000 in Q15 format and corresponds to approximately a 70-Hz bandwidth. After the receiver is locked, this value changes to 0.91 (>7500 in Q15 format), which corresponds to a bandwidth of approximately 10 Hz. The criterion used for the receiver being locked-on is the magnitude of the error function calculated by (32) being less than the threshold used for the carrier recovery (TRSHD1).

Baud Clock

The output of the loop filter, designated by e'_b in Figure 19, drives the local baud clock. The baud clock tracks the sample count (SAMPLE) and thus informs:

1. The decision block when it is the middle of the baud (sample 7) and thus the optimum time for demodulation, and
2. The error signal generator when the sample count is 3 and 11 so that the error signal e_b can be calculated.

These two objectives are achieved with the use of a 16-entry table in the program memory. Each table entry contains the address of a subroutine task to be performed between two consecutive samples. The tasks are numbered 0, 1,..., 15. Table 6 shows the memory map of the 16 tasks performed by the modem receiver.

Table 6. Memory Map of Tasks Performed by the Modem Receiver

```
*
*  TASK MASTER SEQUENCE TABLE  (RECEIVE)
*  TASKS ARE EXECUTED FROM BOTTOM TO TOP
*
TSKSEQ  EQU    $
        DATA   DUMMY    * UNUSED CYCLE                        15
        DATA   DUMMY    * UNUSED CYCLE                        14
        DATA   DUMMY    * UNUSED CYCLE                        13
        DATA   DUMMY    * UNUSED CYCLE                        12
        DATA   BDCLK2   * COMPUTE ENERGY E(11)                11
        DATA   DUMMY    * UNUSED CYCLE                        10
        DATA   OUT      * COMMUNICATE WITH TMS7742             9
        DATA   DECODE   * DECODE/GET SCRAMBLED DIBIT           8
        DATA   DEMODB   * DEMODULATE IN MIDDLE OF BAUD         7
        DATA   DUMMY    * UNUSED CYCLE                         6
        DATA   AGCUPT   * UPDATE THE AGC EVERY 3RD BAUD        5
        DATA   DUMMY    * UNUSED CYCLE                         4
        DATA   BDCLK1   * COMPUTE ENERGY E(3)                  3
        DATA   DUMMY    * UNUSED CYCLE                         2
        DATA   DUMMY    * UNUSED CYCLE                         1
        DATA   DUMMY    * UNUSED CYCLE                         0
```

Task 3 (BDCLK1) calculates the signal energy E(3) (PENRGY). Task 5 updates (once every three bauds) the automatic gain control value. Task 7 (DEMODB) implements the demodulation in the middle of the baud. Task 8 (DECODE) makes the channel decisions based on the demodulated (from Task 7) I and Q values, and decodes the decisions to obtain the scrambled dibits. Task 9 (OUT) performs the data exchange between the TMS32010 and the TMS7742. Task 11 calculates the signal energy E(11) (ENRGY). The TMS32010 code used to drive the table of the modem receiver tasks is shown below.

```
*
*  RECEIVER TASK SEQUENCE DRIVER ROUTINE
*
        LAC   SAMPLE   * DECREMENT THE SAMPLE COUNT
        SUB   ONE      * TO CHECK FOR END OF BAUD
        BGEZ  OVRSAM   * IF NOT, THEN SKIP COUNT RESET
        LACK  15       * RESTART THE SAMPLE COUNTER AT 15
```

```
OVRSAM   SACL  SAMPLE   * SAVE NEW COUNT VALUE
         LACK  TSKSEQ   * GET ADDRESS OF TOP OF TABLE
         ADD   SAMPLE   * ADD IN OFFSET
         TBLR  TEMP     * GET THE PROGRAM ADDRESS
         LAC   TEMP     * FOR THE TASK CALL
         CALA           * EXECUTE THE APPROPRIATE TASK
```

Initially, the sample count (SAMPLE) contains the task number of the previous task performed. This number is decremented so that the next task in the sequence is performed. If the sample count becomes negative, it is reset to 15. The sample count is then added to the address of the top of the task table (TSKSEQ). With the use of the TBLR instruction, the table entry is transferred to the data memory. Each table entry is the address of the subroutine task to be performed. Using the CALA instruction, the equivalent of the 'computed GOTO' used in FORTRAN, the program control transfers to the selected subroutine. For a 9.6-kHz sampling rate, the TMS32010 with a 200-ns cycle time has 512 cycles available to implement each of these tasks. This number of cycles is more than enough since the worst-case task takes approximately 300 cycles. Also, since only 6 out of the 16 tasks are used, 10 more tasks are available for the designer to incorporate additional functions such as an adaptive equalizer, scrambling/descrambling, and synchronous-to-asynchronous and asynchronous-to-synchronous conversions.

The algorithm of adjusting the baud clock based on the filtered error signal e'_b (BEROUT) is the same as the one described earlier for the unfiltered error signal e_b (BERROR), and is summarized below.

$$\begin{array}{ll} e'_b > 0 & \text{delay local baud clock} \\ e'_b < 0 & \text{advance local baud clock} \end{array} \tag{43}$$

The advance or delay of the baud clock is implemented by changing the sample count (SAMPLE) appropriately. In the case of delaying the clock, the middle of the local baud clock (sample 7) occurs earlier than the middle of the incoming baud. Geometrically, sample 7 is located on the left side of the 'energy hill' of Figure 20 instead of at the top. If the sample count does not change, then 16 samples later, sample 7 of the next local baud will again be located on the left side of the 'energy hill' of the next incoming baud. Therefore, the sample count must be decremented by one. Instead of 16 samples, the middle of the next baud is taken to be 17 samples later. Hopefully then, the middle of the local baud is on or at least closer to the top of the 'energy hill.'

The case of advancing the clock is similar, except that the sample count is incremented by one, and thus the middle of the next baud is taken 15 samples after the middle of the current baud.

Clock Recovery Threshold

One more issue, the clock recovery threshold, is associated with the alignment of the baud clock. Since there is a finite number of samples in each baud interval, the baud clock has a finite resolution. Therefore, if the middle of the local baud (sample 7) is within one sample of the middle of the incoming baud, no correction must be applied. A threshold can be used so that corrections are applied only if the magnitude of the filtered error signal is greater than the threshold value. An initial estimate of this threshold is obtained by computing the magnitude of the error signal that corresponds to a one-sample change in the local baud clock. Consider the effect of a one-sample change in Figure 20. The middle of the local baud clock E(7) is translated to E(6) (or E(8)); E(3) is translated to E(2) (or E(4)); and E(11) is to E(10) (or E(12)). Therefore, a one sample change results in an error signal e_b given by (41) of magnitude δ as indicated in Figure 20. Approximating the 'energy hill' with the positive half of a sine wave (see Figure 20), results in $\delta = 0.12$. This would be the threshold if the gain of the clock recovery loop filter were unity. For DC, the gain of this filter is (see equation (37))

$$G_2 = |H_2|\max = \frac{B_2}{1 - A_2} \qquad \text{where } 0 < A_2 < 1$$

The value chosen for the coefficient B_2 (BPLL2) is 0.0024 or >50 in Q15 format. After the receiver is locked-on to the incoming carrier, the coefficient A_2 (BPLL1) is 0.91. The gain G_2 of the loop filter is computed to be $G_2 = 0.026$.

The gain G_2 results in an 'effective threshold' of $\delta = 0.00312$. This corresponds to >33 in Q14 format used for the clock recovery threshold by designer's choice. However, this is just an initial estimate since the mathematical model used is only an approximation. After this estimate was obtained, the final value of the clock recovery threshold (TRSHD2), >8, was determined by trial and error.

The calculation of the thresholds for both the clock and carrier recoveries was performed based on the DC gain of the loop filters. A reason why the calculated thresholds are greater than those obtained by trial and error is that the filter gain is maximum at DC.

Just as in the carrier recovery, a two-level correction is used for the baud clock. If the magnitude of the error signal is less than the threshold, no correction is applied. If the magnitude of the error signal (BERROUT) is greater than the threshold (TRSHD2) but less than twice the threshold, the baud clock is advanced or delayed by one sample. If the magnitude of the error is greater than twice the threshold, then the baud clock is adjusted by two samples.

Functions Implemented in the TMS7742

The Texas Instruments TMS7742 is a microcomputer with an on-chip UART and 4K bytes of internal EPROM. It was included in the modem design to increase its flexibility and upgradability. With the use of the TMS7742, both serial and parallel interfaces with the DTE can be efficiently implemented. The TMS7742 can also perform some of the modem functions, thus allowing the TMS32010 to do more complicated tasks. This flexibility allows the hardware design to be upgradable to 2400-bps splitband modems

(V.22 bis). The TMS7742 acts as a modem controller and performs the asynchronous-to-synchronous and synchronous-to-asynchronous data conversions. It also scrambles the data from the DTE and descrambles the decoded dibits received from the TMS32010 before sending them to the DTE. The TMS7742 code is given in Appendix E.

Asynchronous-to-Synchronous and Synchronous-to-Asynchronous Conversions

Asynchronous data received from the DTE may include a start bit, seven or eight data bits, and one or more stop bits. When the DTE is not sending any data, the modem must still continue to transmit scrambled marks. Even though the DTE can send faster than 1200 bits per second, the modem must transmit only 1200 bits per second to the telephone line. This means that the modem must delete some of the bits received from the DTE. The Bell 212A protocol permits deleting one stop bit every nine characters. The data received from the TMS32010 demodulator may have characters with a deleted stop bit. The TMS7742 must detect the deleted stop bit and add it to the character before sending it to the DTE. The TMS7742 assembles the descrambled dibits into a character, checks for missing stop bits, and adds the missing stop bit if detected. The speed of the UART is set to enable inserting one stop bit in every nine characters; i.e., when transmitting 10 bits per character, adding one bit in nine characters (a total of 90 bits) should not change the speed. Thus, the UART is set to 1/90th of a bit interval faster.

Scrambler/Descrambler

The data that has been converted into synchronous dibits is scrambled using equation (2), which is repeated below.

$$d_S(n) = d(n) \text{ XOR } d_S(n-14) \text{ XOR } d_S(n-17)$$

The TMS7742 holds the previous 17 scrambler outputs in its internal registers and uses the XOR instruction to exclusively-OR the proper bits to generate the new scrambled output. After scrambling each bit, these registers are shifted by one and saved to provide the $(n-7)$ outputs for the next bit.

A similar routine is used to descramble the decoded data received from the TMS32010. The descrambling is performed using equation (3) repeated below.

$$d(n) = d_S(n) \text{ XOR } d_S(n-14) \text{ XOR } d_S(n-17)$$

Performance

The performance of the modem implemented using the TMS32010 was evaluated using automatic modem testing equipment. A block diagram of this testing equipment is shown in Figure 21.

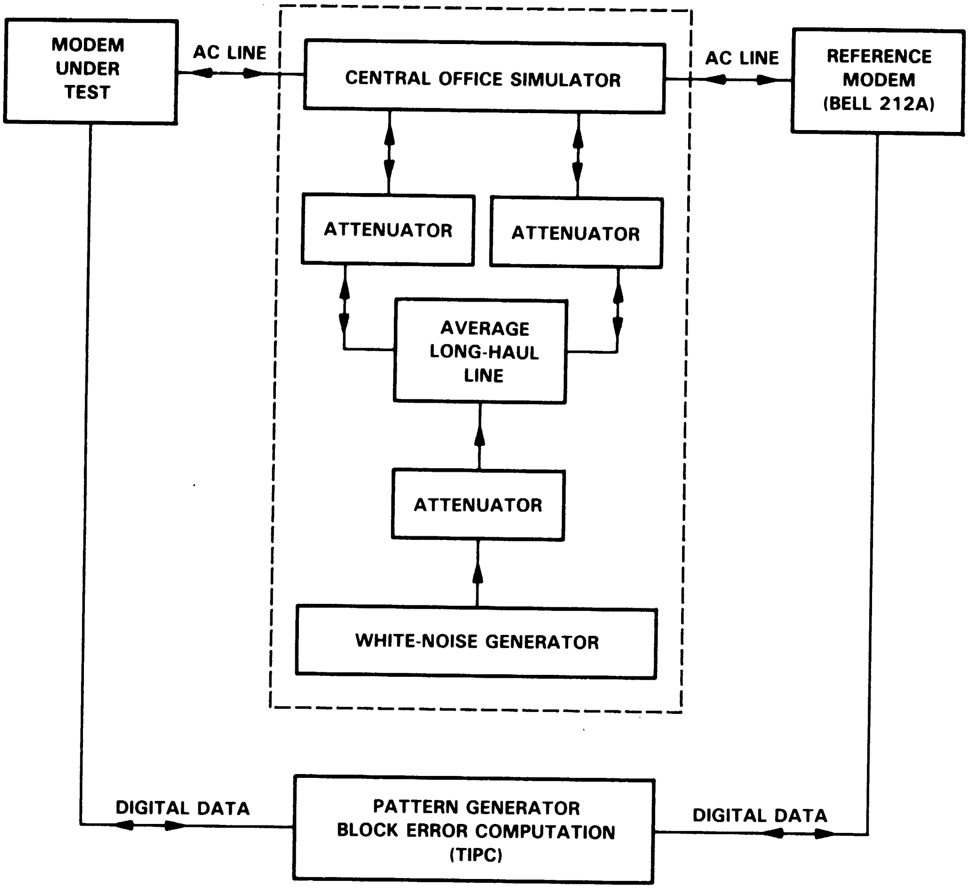
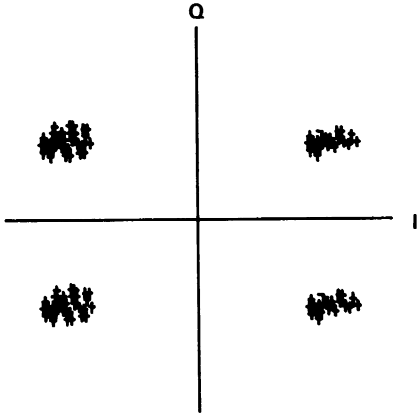


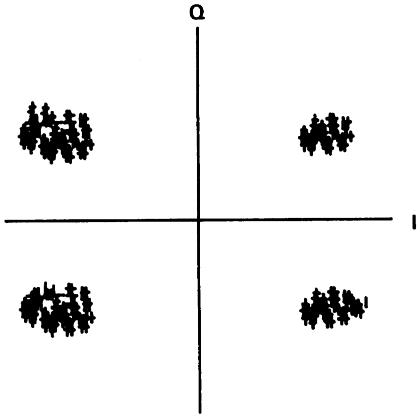
Figure 21. Modem Testing Equipment

The testing environment in Figure 21 provides a Central Office simulator, an average long-haul line simulator, and a C-notched white-noise generator. The attenuators provide signal-level and noise-level attenuation. The testing is performed under full-duplex and maximum data throughput conditions.

The average long-haul line effects are evident from the differences between the signal constellation diagrams of Figures 22(a) and 22(b). Figure 22(a) shows the signal constellation with the TMS32010 modem in the analog loop-back mode. Figure 22(b) shows the signal constellation with the TMS32010 modem operating over an average long-haul line at a 14-db signal-to-noise ratio. The presence of the average long-haul line results in a 'spreading' of the signal constellation points. This spreading implies a higher probability of error since the signal points used to make the decisions approach the decision boundaries.



(a) SIGNAL CONSTELLATION IN ANALOG LOOP-BACK MODE



(b) SIGNAL CONSTELLATION OVER AVERAGE LONG-HAUL LINE

Figure 22. Signal Constellation Diagrams

Referring to Figure 21, the Texas Instruments Professional Computer generates random characters. These characters are sent to the reference modem and the modem under testing. The modems transmit the characters they receive to each other, and each modem sends the characters received to the Professional Computer. The computer then compares the received characters with the ones originally created to determine the error rate. The error rate is determined in terms of percent error-free blocks. Each block consists of 512 characters (5120 bits) and is considered to be error-free only if all of the bits in the block are received with no error.

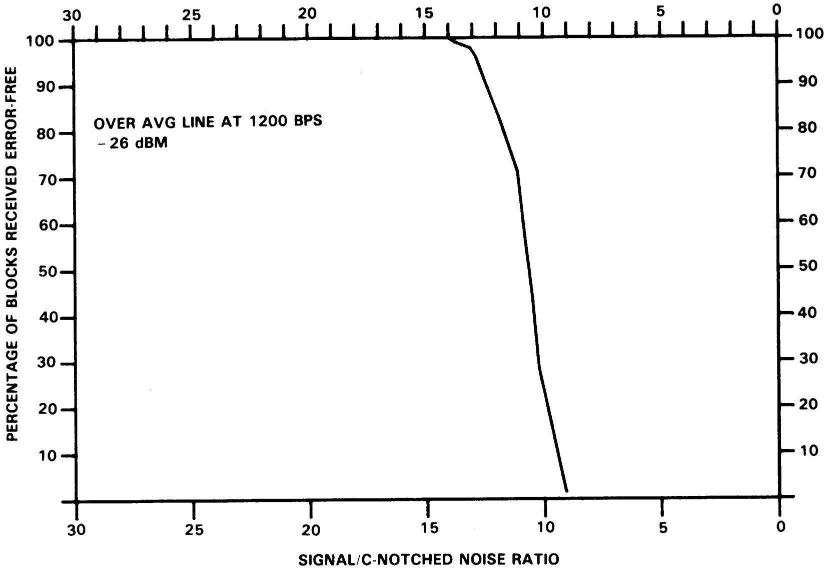
In all the tests performed, the Bell 212A modem was the reference modem configured in the answer mode. The reason for this is that only an originate

TMS32010-based modem is implemented. The answer mode is not included because, as mentioned in the "Introduction," this is beyond the purpose of this report. To incorporate the answer mode, two tables must be added in the TMS32010 code presented in Appendix D. The first table should contain the coefficients of the two receiver input bandpass filters with a passband centered around 1200 Hz. The second table should contain the increments used by the sine-table driver routine so that a 2400-Hz carrier is generated for the transmitter and a 1200-Hz carrier is generated for the receiver. When the TMS7742 configures the TMS32010 in the answer mode, the filter coefficients and the sine-table increments can be transferred from the program memory to the data memory with the use of the TBLR instruction. No performance difference is expected between the answer and originate modes.

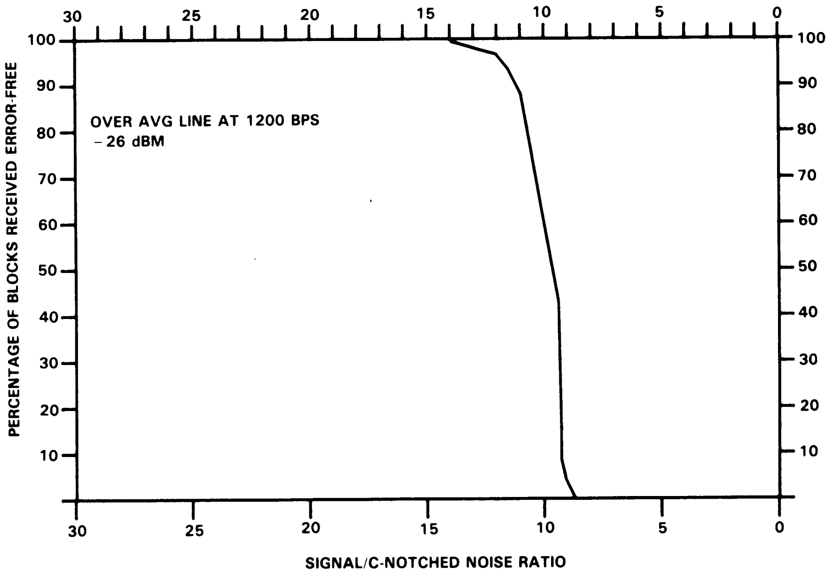
In Figure 23, the vertical axis indicates the percentage of blocks received error-free and the horizontal axis is the signal-to-noise ratio in db. The percentage of error-free blocks is calculated at each signal-to-noise ratio level (30, 29, 28,...) based on the number of error-free blocks received out of 1024 transmitted. All tests were performed at a −26 dbm (0.1 V) signal level. Figure 23(a) shows the test results with the TMS32010-based modem as the modem under testing. The vertical axis of Figure 23(a) is the percentage of blocks received error-free by the Bell modem. Figure 23(b) shows the results when the AT&T Dataphone II is used instead of the TMS32010-based modem.

Since the Bell modem is used as a reference modem, the above results indicate how well the transmitters of the TMS32010-based modem and the AT&T modem are performing. From Figures 23(a) and 23(b), it can be seen that for both the TMS32010 and AT&T modems, block errors start occurring at a signal-to-noise ratio of approximately 13 db and that the curve corresponding to the TMS32010 modem falls slightly faster. Therefore, the performance of both modem transmitters is approximately the same with the AT&T transmitter performing slightly better than the TMS32010 transmitter. Figure 24(a) shows the percentage of blocks received error-free by the TMS32010-based modem. The Bell modem (reference modem) is used to transmit these blocks. Figure 24(b) shows the percentage of blocks received error-free by the AT&T modem with the Bell modem transmitting.

It can be seen that the AT&T receiver performs approximately 2 db better than the TMS32010 receiver. The performance of the TMS32010 modem receiver could be improved with the inclusion of more filter taps in the receiver input bandpass filters.

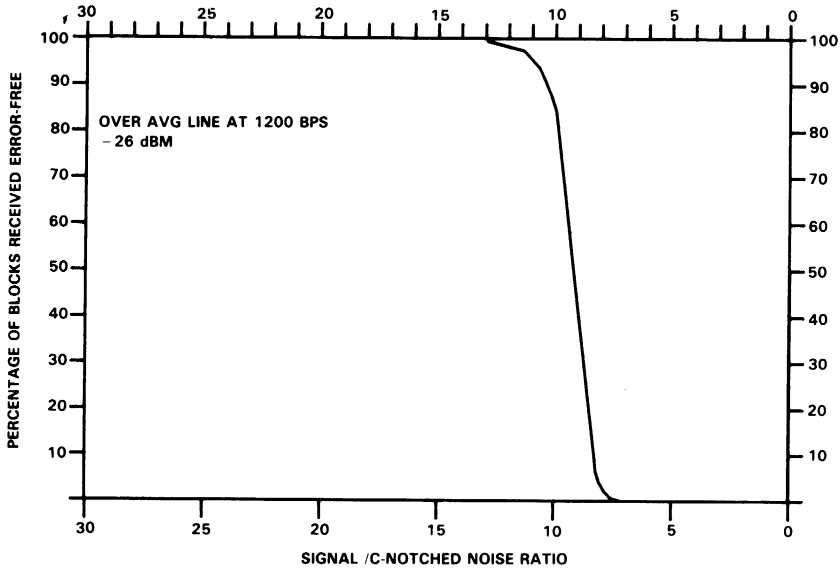


(a) PERCENTAGE OF BLOCKS RECEIVED ERROR-FREE BY THE BELL 212A MODEM VS. SNR WITH THE TMS32010 MODEM ORIGINATING

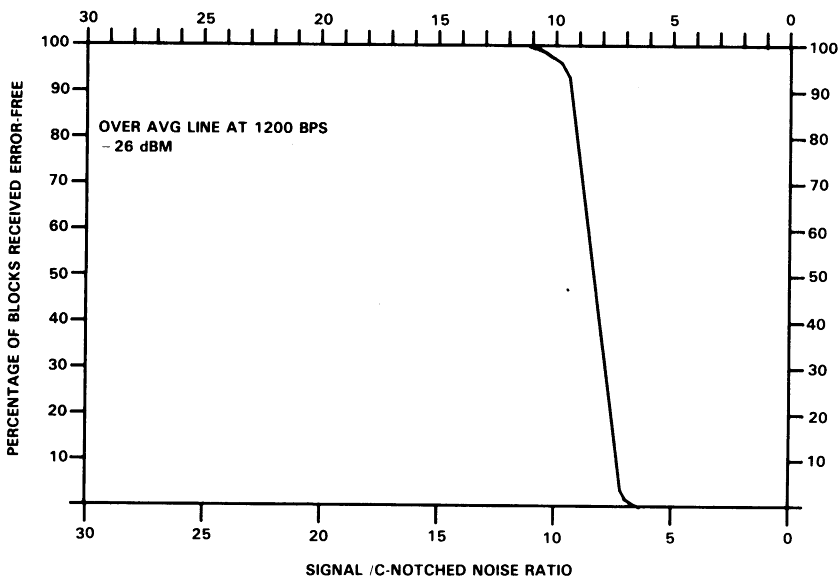


(b) PERCENTAGE OF BLOCKS RECEIVED ERROR-FREE BY THE BELL 212A MODEM VS. SNR WITH THE AT&T MODEM ORIGINATING

Figure 23. Performance of TMS32010 and AT&T Modem Transmitters



(a) PERCENTAGE OF BLOCKS RECEIVED ERROR-FREE BY THE TMS32010 MODEM VS. SNR WITH THE BELL MODEM TRANSMITTING



(b) PERCENTAGE OF BLOCKS RECEIVED ERROR-FREE BY THE AT&T MODEM VS. SNR WITH THE BELL MODEM TRANSMITTING

Figure 24. Performance of TMS32010 and AT&T Modem Receivers

Other Implementation Considerations

The implementation approach of the Bell 212A/V.22 modem presented in the previous sections is not unique. There are other and possibly more efficient ways of implementing the modem.

Drastic reduction of the hardware cost results from the use of a codec for the A/D and D/A conversions instead of the 12-bit linear A/D and D/A converters used in this implementation. This approach becomes even more attractive with the use of the TMS32011 digital signal processor in place of the TMS32010. The TMS32011 is a microcomputer (no external memory expansion) having the same architecture as the TMS32010 with the additional feature of containing the necessary logic for interfacing to a codec. In this implementation, the necessary input bandpass filtering for the modem receiver can be performed with an AMI S35212A analog filter chip. The modem hardware block diagram of this implementation is shown in Figure 25.

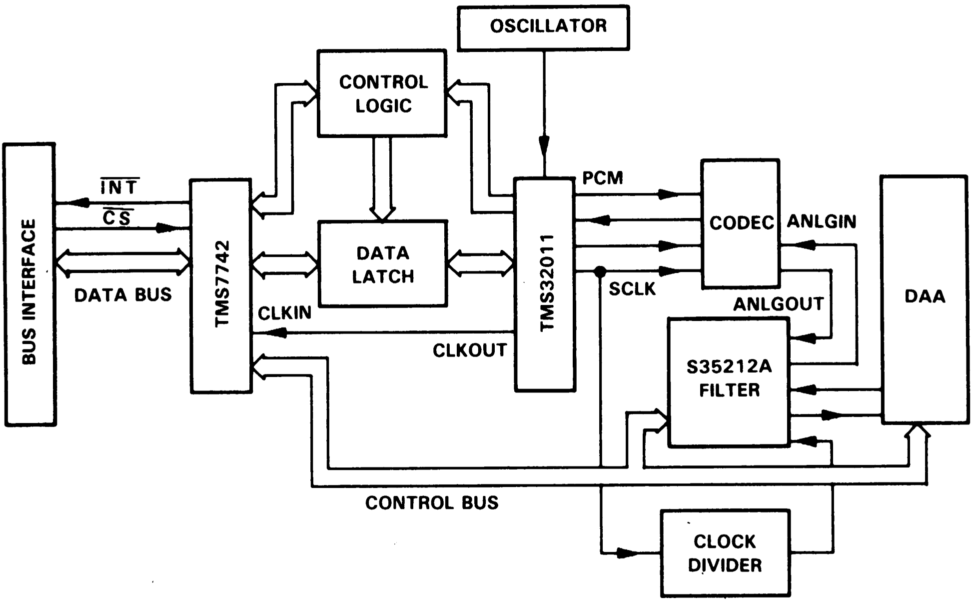


Figure 25. Modem Hardware Block Diagram Using a Codec for the A/D and D/A Conversions

If this approach is used, the receiver input has the configuration shown in Figure 26. The bandpass filtering is implemented in the analog domain and the Automatic Gain Control and Hilbert Transformer Pair implemented in the digital domain inside the TMS32011. Implementing the bandpass filtering in the analog domain should save adequate program memory, data memory, and processing power to allow the design to be upgraded to the V.22 bis specification. If only the Bell 212A is of interest, the bandpass filtering could be performed digitally within the TMS32011.

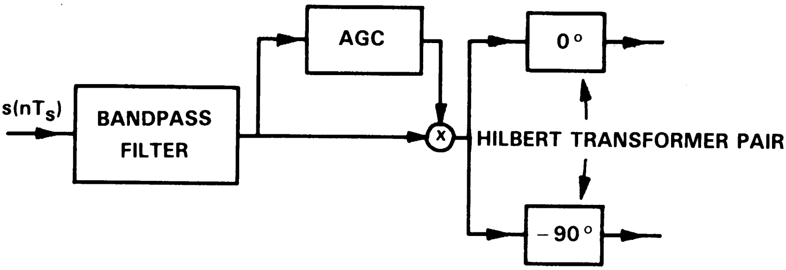


Figure 26. Alternative Modem Receiver Input Configuration

Conclusions

This application report discussed the digital implementation of splitband modems using the TMS32010 general-purpose high-speed digital signal processor. The theory and implementation of the Bell 212A/V.22 full-duplex modem was covered in detail. With a modification of some of the functional blocks of the Bell 212A/V.22, 2400-bps splitband modems (V.22 bis) can be implemented.

Modems are sophisticated devices, consisting of many functional blocks. This implies the need of special features for the microprocessor to be used. The TMS32010 with a 200-ns cycle, an on-board single-cycle multiplier, and a special instruction set tailored for digital signal processing is able to implement the modem functional blocks (see Table 5) with approximately 60-percent use of the available processing power. The modem program utilizes 103 words of data memory out of the 144 words available. This corresponds to approximately 71 percent of the data memory. The program also utilizes 954 words of program memory out of the 1536 words available, corresponding to approximately 62 percent of the on-chip program memory. Therefore, the use of the full-speed off-chip memory feature of the TMS32010 was not utilized. Since a large portion of the power of the TMS32010 is still available, additional functions, such as an adaptive equalizer and the Data Encryption Standard (DES)[1], can be implemented with the inclusion of new code. With a 6-percent loading of the TMS32010, the DES can provide secure communication between 1200-bps full-duplex modems.

The TMS32010 is one of many digital signal processors in the TMS320 family. The flexibility and processing power of the TMS320 family provide high performance, high reliability, and cost-effective solutions for medium- and high-speed modems.

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Appendix A

Derivation of Demodulator Structure Equations

The equations that describe the demodulator structure (see Figure 6) of the modem receiver are derived in this Appendix. The background material required for this derivation is presented first. The following discussion requires a basic knowledge of complex variables.

The baseband signal, at the output of the transmitter digital lowpass filters (see Figure 3), can be expressed as a complex value

$$c(nT_S) = I(nT_S) - j\,Q(nT_S) \tag{A-1}$$

For transmission through the telephone network, this signal is modulated to the voice frequencies. Modulation involves multiplication by a complex exponential.[18] The modulated signal is then given by

$$m(nT_S) = c(nT_S)\,e^{j\omega_c nT_s} \tag{A-2}$$

where ω_c is the carrier frequency. Substitution of (A-1) into (A-2), and the use of the identity

$$e^{j\omega_c nT_s} = \cos(\omega_c nT_S) + j\,\sin(\omega_c nT_S)$$

give

$$m(nT_S) = \{I(nT_S)\cos(\omega_c nT_S) + Q(nT_S)\sin(\omega_c nT_S)\}$$
$$+\ j\,\{I(nT_S)\sin(\omega_c nT_S) - Q(nT_S)\cos(\omega_c nT_S)\} \tag{A-3}$$

The real and imaginary parts of (A-3) are later shown to be a Hilbert transform pair. Two signals are referred to as a Hilbert transform pair if they are related with a Hilbert transform. A Hilbert transform is implemented with a filter called a Hilbert transformer. The Hilbert transform pair property that relates the real and imaginary parts of (A-3) allows the transmission of the real part of (A-3) only. The imaginary part is recovered at the receiver by Hilbert transforming the incoming signal. Figure A-1 shows the spectrum of the complex baseband information $c(nT_S)$. Figure A-2 shows the spectrum after modulation by the complex exponential (see equation (A-2)). This is the spectrum of $m(nT_S)$. Figure A-3 shows the spectrum of the transmitted signal, i.e., the spectrum of the real part of $m(nT_S)$.

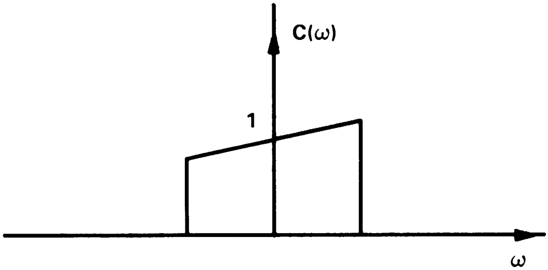

Figure A-1. Spectrum of Complex Baseband Information

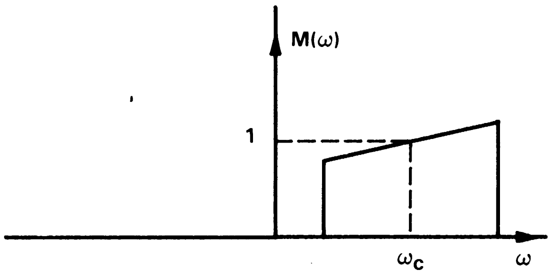

Figure A-2. Spectrum after Modulation

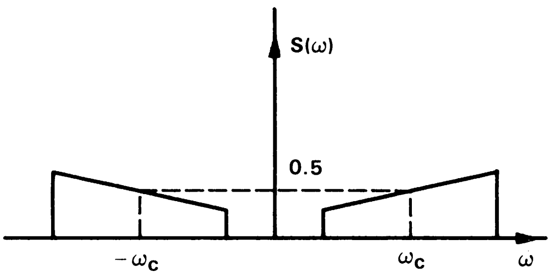

Figure A-3. Transmitted Spectrum

A Hilbert transformer is defined to be a filter with the transfer function[18]

$$H_t(\omega) = -e^{j\frac{\pi}{2}}\,\mathrm{sgn}(\omega) = -j\ \mathrm{sgn}(\omega)\ \text{(A-4)} \qquad \text{(A-4)}$$

where sgn is the sign function defined by equation (29). The transfer function characteristics of the Hilbert transformer are shown in Figure A-4, where it is seen that the Hilbert transformer introduces a -90 degree phase shift for positive frequencies ($\omega > 0$), and a $+90$ degree phase shift for negative frequencies ($\omega < 0$).

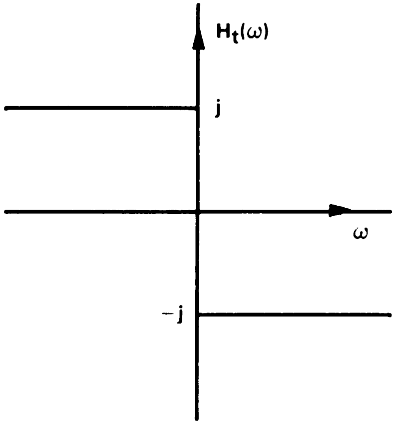

Figure A-4. Hilbert Transformer Transfer Function

The Hilbert transform pair relationship between the real and imaginary parts of (A-3) is discussed next. It is shown that the imaginary part of $m(nT_S)$ is the output of a Hilbert transformer with the input being the real part of $m(nT_S)$. The analysis is performed in the frequency domain where multiplication is replaced by convolution. Let $S(\omega)$ and $\hat{S}(\omega)$ be the Fourier transforms of the real and imaginary parts of $m(nT_S)$, respectively. Then (see equation (A-3))

$$S(\omega) = \frac{1}{2}\{I(\omega - \omega_C) + I(\omega + \omega_C)\} + \frac{j}{2}\{Q(\omega + \omega_C) - Q(\omega - \omega_C)\} \tag{A-5}$$

$$\hat{S}(\omega) = \frac{j}{2}\{I(\omega + \omega_C) - I(\omega - \omega_C)\} - \frac{1}{2}\{Q(\omega - \omega_C) + Q(\omega + \omega_C)\} \tag{A-6}$$

where $I(\omega)$ and $Q(\omega)$ are the Fourier transforms of $I(nT_S)$ and $Q(nT_S)$, respectively.

With $S(\omega)$ as the input to the Hilbert transformer (transfer function $H_t(\omega)$), the output in the frequency domain is given by

$$O(\omega) = S(\omega)\, H_t(\omega) = -j\, S(\omega)\, \mathrm{sgn}(\omega) \tag{A-7}$$

Substitution of (A-5) into (A-7) gives

$$\begin{aligned} O(\omega) = {} & -j\,\Big\{\frac{1}{2}\{I(\omega - \omega_C) + I(\omega + \omega_C)\} \\ & + \frac{j}{2}\{Q(\omega + \omega_C) - Q(\omega - \omega_C)\}\Big\}\, \mathrm{sgn}(\omega) \end{aligned} \tag{A-8}$$

Since for positive frequencies ($\omega > 0$),

$$I(\omega + \omega_C) = 0$$

$$Q(\omega + \omega_C) = 0 \tag{A-9}$$

and for negative frequencies ($\omega < 0$),

$$\begin{aligned} I(\omega - \omega_C) &= 0 \\ Q(\omega - \omega_C) &= 0 \end{aligned} \tag{A-10}$$

equation (A-8) simplifies to

$$O(\omega) = \begin{cases} -\frac{j}{2} I(\omega - \omega_C) - \frac{1}{2} Q(\omega - \omega_C) & \text{where } \omega > 0 \\ \frac{j}{2} I(\omega + \omega_C) - \frac{1}{2} Q(\omega + \omega_C) & \text{where } \omega < 0 \end{cases} \tag{A-11}$$

Substitution of (A-9) and (A-10) into (A-6) and comparison of the result with (A-11) shows that $S(\omega) = O(\omega)$.

Therefore, the real and imaginary parts of $m(nT_S)$ (see equation (A-3)) represent a Hilbert tranform pair. With $s(nT_S)$ and $\hat{s}(nT_S)$ denoting the real and imaginary parts of m(nTs), respectively, (A-3) can be written as

$$m(nT_S) = s(nT_S) + j\,\hat{s}(nT_S) \tag{A-12}$$

At the receiver end, recovery of the imaginary part $\hat{s}(nT_S)$ involves Hilbert transforming the real part $s(nT_S)$ (incoming signal), as shown in Figure A-5.

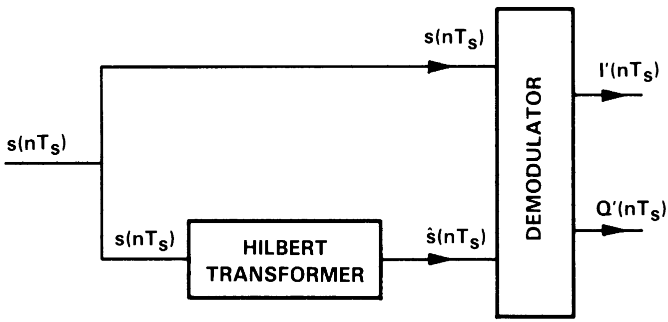

Figure A-5. Recovery of Complex Information by Hilbert Transforming the Incoming Signal

Consider the Fourier transform of (A-12)

$$\begin{aligned} M(\omega) &= S(\omega) + j\,\hat{S}(\omega) \\ &= S(\omega) + j\,\{-j\,S(\omega)\,\mathrm{sgn}(\omega)\} \\ &= S(\omega) + S(\omega)\,\mathrm{sgn}(\omega) \end{aligned}$$

Therefore,

$$M(\omega) = \begin{cases} 2\,S(\omega) & \text{where } \omega > 0 \\ 0 & \text{where } \omega < 0 \end{cases} \tag{A-13}$$

i.e., the spectrum of $m(nT_S)$ is zero for negative frequencies (see Figure A-2). If this property does not hold due to the use of a nonideal Hilbert transformer, harmonics appear at the output of the receiver demodulator (see Appendix B).

The equations that describe the receiver demodulator are derived next. The demodulator translates the recovered complex modulated information back to the baseband. This is accomplished by multiplying the passband information with a complex exponential.

$$c'(nT_S) = m(nT_S)\, e^{-j\omega_C nT_S} \tag{A-14}$$

where $c'(nT_S)$ is the recovered baseband signal, $m(nT_S)$ is the passband signal given by (A-12), and ω_C is the carrier frequency recovered at the receiver by the carrier recovery algorithm.

Substitution of (A-12) into (A-14) gives

$$\begin{aligned} c'(nT_S) = {} & \{s(nT_S)\cos(\omega_C nT_S) + \hat{s}(nT_S)\sin(\omega_C nT_S)\} \\ & + j\,\{\hat{s}(nT_S)\cos(\omega_C nT_S) - s(nT_S)\sin(\omega_C nT_S)\} \end{aligned} \tag{A-15}$$

The complex baseband information $c'(nT_S)$ is also given by (see equation (A-1) and Figure 5)

$$c'(nT_S) = I'(nT_S) - j\,Q'(nT_S) \tag{A-16}$$

Equating the real and imaginary parts of (A-15) to those of (A-16) results in

$$I'(nT_S) = s(nT_S)\cos(\omega_C nT_S) + \hat{s}(nT_S)\sin(\omega_C nT_S) \tag{A-17}$$

$$Q'(nT_S) = s(nT_S)\sin(\omega_C nT_S) - \hat{s}(nT_S)\cos(\omega_C nT_S) \tag{A-18}$$

Equations (A-17) and (A-18) describe the receiver demodulator of Figure 6.

Appendix B
Effects of Nonideal Hilbert Transformers

The effect of nonideal Hilbert Transformers in modem design is studied in this Appendix. The following discussion requires a basic knowledge of complex variables.

The nonideal Hilbert transformer characteristics differ from the ideal ones shown in Figure 28 and described by equation (A-4) in Appendix A. The phase shift introduced by the nonideal filter is not exactly 90 degrees. The transfer function characteristics of such a filter are given by

$$H'(\omega) = -e^{j\left(\frac{\pi}{2}+\alpha\right)} \operatorname{sgn}(\omega) = -j\, e^{j\alpha} \operatorname{sgn}(\omega) \tag{B-1}$$

where 'α' is a nonzero constant indicating the deviation from the ideal filter.

Consider the effect of a nonideal Hilbert transformer described by equation (B-1). The incoming signal $s(nT_S)$ is the real part of $m(nT_S)$. This signal is filtered by the nonideal Hilbert transformer to generate at the output a signal $\hat{s}'(nT_S)$ different from $\hat{s}(nT_S)$ (see Appendix A). With $\hat{S}'(\omega)$ as the Fourier transform of $\hat{s}'(nT_S)$, the output of the nonideal Hilbert transformer can be described in the frequency domain by

$$\hat{S}'(\omega) = H'(\omega)\, S(\omega) = -j\, e^{j\alpha} \operatorname{sgn}(\omega)\, S(\omega) \tag{B-2}$$

The complex signal at the input of the receiver demodulator is described by

$$m'(nT_S) = s(nT_S) + j\, \hat{s}'(nT_S) \tag{B-3}$$

The frequency-domain equivalent of (B-3) is

$$M'(\omega) = S(\omega) + j\, \hat{S}'(\omega) \tag{B-4}$$

Substitution of (B-2) into (B-4) gives

$$M'(\omega) = S(\omega) + e^{j\alpha} \operatorname{sgn}(\omega)\, S(\omega) \tag{B-5}$$

Equation (B-5) can be written as

$$M'(\omega) = \begin{array}{ll} S(\omega)\,\{\,1 + e^{j\alpha}\,\} & \text{where } \omega > 0 \\ S(\omega)\,\{\,1 - e^{j\alpha}\,\} & \text{where } \omega < 0 \end{array} \tag{B-6}$$

For a nonideal Hilbert transformer, the parameter 'α' is nonzero. This results in $M'(\omega)$ having nonzero components at negative frequencies as indicated by (B-6). The spectrum of the signal at the input of the receiver demodulator is shown in Figure B-1. Comparison of Figures A-2 and B-1 indicates that the effect of the nonideal Hilbert transformer is the generation of nonzero spectral components at negative frequencies.

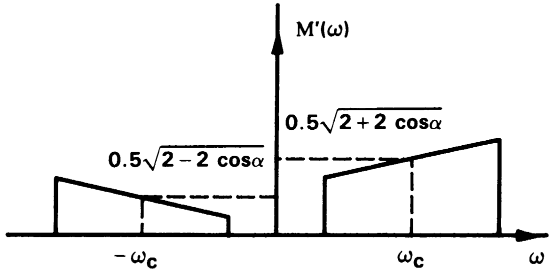

Figure B-1. Effect of Nonideal Hilbert Transformer on the Spectrum of the Complex Signal at the Input of the Demodulator

The effect of the receiver demodulator on the spectrum of Figure B-1 is shown in Figure B-2.

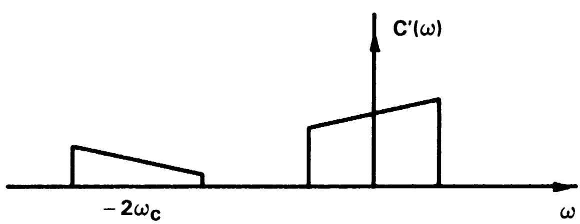

Figure B-2. Effect of Nonideal Hilbert Transformer on the Spectrum of the Baseband Complex Signal

Figure B-2 indicates that harmonics appear at the output of the demodulator. The frequency of these harmonics is twice the carrier frequency. Their elimination involves the use of lowpass filters at the output of the demodulator. These filters, however, introduce group delay and possibly phase delay effects that affect the carrier recovery and decision algorithms. Compensation for the lowpass filter side-effects results in a more complicated modem receiver design. Such nonideal Hilbert transformers are encountered in analog modems. This appendix has demonstrated one more advantage of a digital implementation of a modem using the TMS32010 digital signal processor.

Appendix C
Automatic Gain Control Table Generator Code

```
10 '       THIS PROGRAM GENERATES THE GAIN TABLE FOR THE AUTOMATIC
20 '       GAIN CONTROL ALGORITHM IN THE MODEM CODE
30 '
40 '       THE PROGRAM PROMPTS THE USER IN THE FOLLOWING MANNER:
50 '
60 '             AGC TABLE ADJUST FACTOR ?
70 '              This feature allows the AGC to gain to a level lower
80 '              than unity.  The entry for unity gain is 256, to set
90 '              the gain lower than unity enter the appropriate per-
100 '              centage of 256.
110 '
120 '            ENTER NAME OF OUTPUT FILE =
130 '              This prompt request the name of a MSDOS format file
140 '              name to store the generated table.
150 '
160 '            TABLE LENGTH =
170 '              This feature allows the user to generate different
180 '              length AGC tables.  This allows the accuracy of the
190 '              table to vary by the number of entries.  The number
200 '              of entries is tied to the number of bits used in the
210 '              table lookup.  In the modem algorithm six bits were
220 '              used in the lookup, therefore the table length will be
230 '              64 words.
240 '
250 '      THE TABLE GENERATED WILL INCLUDE DESCRIPTIVE COMMENTS AND WILL
260 '      BE IN A FORM READY TO BE ADDED DIRECTLY INTO THE ASSEMBLY CODE
270 '      FOR AN ALGORITHM.  SINCE THE AGC SOFTWARE SHIFTS THE LOOKUP
280 '      VALUE TO THE MOST SIGNIFICANT BIT THE FIRST HALF OF THE AGC TABLE
290 '      (THE LESS ACCURATE HALF) WILL NOT BE USED.  THEREFORE THE USER
300 '      CAN DELETE THE FIRST HALF AND SAVE A CONSIDERABLE AMOUNT OF PROGRAM
310 '      MEMORY SPACE.
320 '
330 '      THIS PROGRAM WAS WRITTEN BY PETER EHLIG FOR USE ON A
340 '        TEXAS INSTRUMENTS PROFESSIONAL COMPUTER
350 '      THE CODE TO MY KNOWLEDGE IS WRITTEN IN STANDARD MS-BASIC AND
360 '      SHOULD OPERATE ON ANY MSDOS SYSTEM.
370 '
380 PRINT 'PROGRAM STARTED"
390 DIM TBLD(500),HTB$(500)
400 OPEN "LPT1:" FOR OUTPUT AS #1
410 INPUT "AGC TABLE ADJUSTMENT FACTOR ? ",GAINADJ
420 INPUT "ENTER NAME OF OUTPUT FILE = ",OUTFILE$
430 OPEN OUTFILE$ FOR OUTPUT AS #3
440 PI = 3.1415927#
450 PI2 = PI * 2
460 INPUT "TABLE LENGTH = ",TBLEN
470 GOSUB 820  ' GENERATE TABLE HEADER
480 DELTA = 327681 / TBLEN
490 FOR I = 1 TO TBLEN
500 TBL = INT(32767 / (I * DELTA) * GAINADJ)
510 TBLD(I) = TBL
520 HTBL$ = HEX$(TBL)
530 HTB$(I) = HTBL$
540 GOSUB 690  ' DISPLAY RANGE ACCURACY  (OPTIONAL)
550 NEXT
560 GOTO 650
570 ' SAVE AGC TABLE TO DISK
580 PRINT#3, "         DATA ";
590 PRINT#3, USING ">\  \";HTB$(I);
600 PRINT#3, "                    ";
610 TBLD1 = TBLD(I) / 256
620 PRINT#3, USING "###.#######";TBLD1
630 RETURN
640 ' END OF AGC TABLE SAVE ROUTINE
650 GOSUB 940  ' DISPLAY SECOND LEVEL LOOKUP
660 GOSUB 880  ' GENERATE TABLE TERMINATION COMMENTS
670 PRINT "PROGRAM FINISHED"
680 END
```

```
690 ' THIS ROUTINE DISPLAYS INFORMATION ABOUT THE RANGE
700 ' ACCURACY OF EACH STEP OF THE TABLE
710 TBLRL = (I - 1) * DELTA - 256
720 IF TBLRL < 0 THEN TBLRL = 0
730 SH1$ = HEX$(TBLRL)
740 SH1A$ = HEX$(TBLRL * TBL / 256)
750 TBLRH = (I - 1) * DELTA + 255
760 SH2$ = HEX$(TBLRH)
770 SH2A$ = HEX$(TBLRH * TBL / 256)
780 PRINT I;TBL;HTBL$;" ";SH1$;" ";SH1A$;" ";SH2$;" ";SH2A$
790 ' PRINT#1,I;TBL;HTBL$;" ";SH1$;" ";SH1A$;" ";SH2$;" ";SH2A$
800 RETURN
810 ' END OF RANGE INFORMATION
820 ' THE ROUTINE GENERATES THE HEADER COMMENTS FOR THE TABLE
830 PRINT#3,"****************************************************************"
840 PRINT#3,"AGCTBL EQU  $             AGC TABLE LENGTH = ";
850 PRINT#3, USING "###";TBLEN
860 RETURN
870 ' END OF HEADER ROUTINE
880 ' THIS ROUTINE GENERATES THE TABLE TERMINATION COMMENTS
890 PRINT#3,"****************************************************************"
900 PRINT#3, "       PAGE"
910 CLOSE
920 RETURN
930 ' END OF TERMINATOR ROUTINE
940 ' TRY SECOND LEVEL LOOKUP
950 DELTA1 = DELTA * 8
960 FOR I = 1 TO 64
970 GOSUB 570  ' SAVE AGC TABLE TO DISK
980 TBLRL = (I - 1) * DELTA - 256
990 IF TBLRL < 0 THEN TBLRL = 0
1000 TBLRH = (I - 1) * DELTA + 255
1010 SH1$ = HEX$(TBLRL)
1020 SH2$ = HEX$(TBLRH)
1030 GOSUB 1100 ' CALCULATE ACCURACY STEPS
1040 SH1A$ = HEX$(TBLRL * TBLD(TBLR1) / SHF1)
1050 SH2A$ = HEX$(TBLRH * TBLD(TBLR2) / SHF1)
1060 PRINT I;TBL;HTBL$;" ";SH1$;" ";SH1A$;" ";SH2$;" ";SH2A$;TBLR1;TBLR2;SHF1
1070 ' PRINT#1,I;TBL;HTBL$;" ";SH1$;" ";SH1A$;" ";SH2$;" ";SH2A$;TBLR1;TBLR2;SHF
1
1080 NEXT
1090 RETURN
1100 'TABLE LOOKUP SHIFTER
1110 TBLEV = TBLRH - 4096
1120 IF TBLEV > 0 GOTO 1180
1130 TBLEV = TBLEV + 2048
1140 IF TBLEV > 0 GOTO 1220
1150 TBLEV = TBLEV + 1024
1160 IF TBLEV > 0 GOTO 1260
1170 GOTO 1300
1180 TBLR1 = I
1190 TBLR2 = I
1200 SHF1 = 256
1210 RETURN
1220 TBLR2 = FIX(TBLRH / 64) + 1
1230 TBLR1 = FIX(TBLRL / 64) + 1
1240 SHF1 = 32
1250 RETURN
1260 TBLR2 = FIX(TBLRH / 32) + 1
1270 TBLR1 = FIX(TBLRL / 32) + 1
1280 SHF1 = 16
1290 RETURN
1300 TBLR2 = FIX(TBLRH / 16) + 1
1310 TBLR1 = FIX(TBLRL / 16) + 1
1320 SHF1 = 8
1330 RETURN
```

The following 64-point table was generated using the 124 for the AGC table adjust factor.

```
************************************************************
AGCTBL EQU  $                AGC TABLE LENGTH =   64
       DATA >1EFF                   30.9961000
       DATA >F7F                    15.4960900
       DATA >A55                    10.3320300
       DATA >7BF                     7.7460940
       DATA >633                     6.1992190
       DATA >52A                     5.1640630
       DATA >46D                     4.4257820
       DATA >3DF                     3.8710940
       DATA >371                     3.4414060
       DATA >319                     3.0976560
       DATA >2D1                     2.8164060
       DATA >295                     2.5820310
       DATA >262                     2.3828130
       DATA >236                     2.2109380
       DATA >211                     2.0664060
       DATA >1EF                     1.9335940
       DATA >1D2                     1.8203130
       DATA >1B8                     1.7187500
       DATA >1A1                     1.6289060
       DATA >18C                     1.5468750
       DATA >179                     1.4726560
       DATA >168                     1.4062500
       DATA >159                     1.3476560
       DATA >14A                     1.2890630
       DATA >13D                     1.2382810
       DATA >131                     1.1914060
       DATA >125                     1.1445310
       DATA >11B                     1.1054690
       DATA >111                     1.0664060
       DATA >108                     1.0312500
       DATA >FF                      0.9960938
       DATA >F7                      0.9648438
       DATA >F0                      0.9375000
       DATA >E9                      0.9101562
       DATA >E2                      0.8828125
       DATA >DC                      0.8593750
       DATA >D6                      0.8359375
       DATA >D0                      0.8125000
       DATA >CB                      0.7929688
       DATA >C6                      0.7734375
       DATA >C1                      0.7539063
       DATA >BC                      0.7343750
       DATA >B8                      0.7187500
       DATA >B4                      0.7031250
       DATA >B0                      0.6875000
       DATA >AC                      0.6718750
       DATA >A8                      0.6562500
       DATA >A5                      0.6445313
       DATA >A1                      0.6289063
       DATA >9E                      0.6171875
       DATA >9B                      0.6054688
       DATA >98                      0.5937500
       DATA >95                      0.5820313
       DATA >92                      0.5703125
       DATA >90                      0.5625000
       DATA >8D                      0.5507813
       DATA >8B                      0.5429688
       DATA >88                      0.5312500
       DATA >86                      0.5234375
       DATA >84                      0.5156250
       DATA >82                      0.5078125
       DATA >7F                      0.4960938
```

```
        DATA >7D                        0.4882813
        DATA >7B                        0.4804688
*********************************************************
        PAGE
```

```
10 '
20 '      This program generates sine table in a format compatible
30 '      to the 320 assembler.  This allows the user to generate
40 '      any length sine table and this program will calculate the
50 '      table entries, configure them in a format compatible to
60 '      the assembler, and document the code.
70 '
80 '      The program prompts the user in the following manner:
90 '
100 '             ENTER NAME OF OUTPUT FILE =
110 '               This prompt request the name of a MSDOS format file
120 '               name to store the generated table.
130 '
140 '             TABLE LENGTH =
150 '               This feature allows the user to select the length of
160 '               the sine table to be generated and therefore the
170 '               accuracy of the table steps.
180 '
190 '
200 '       This program was written by Peter Ehlig for use on a
210 '         Texas Instruments Professional Computer
220 '       The code to my knowledge is written in standard MS-BASIC and
230 '       should operate on any MSDOS system.
240 '
250 PRINT 'PROGRAM STARTED"
260 INPUT "ENTER NAME OF OUTPUT FILE = ",OUTFILE$
270 OPEN OUTFILE$ FOR OUTPUT AS #3
280 PI = 3.1415927#
290 PI2 = PI * 2
300 INPUT "TABLE LENGTH = ",TBLEN
310 DELTA = PI2 / TBLEN
320 INDX1 = -DELTA
330 NETDEG = 360 / TBLEN
340 PRINT#3,"**********************************************************"
350 PRINT#3,"SINE   EQU  $             SINE TABLE LENGTH = ";
360 PRINT#3, USING "###";TBLEN
370 FOR I = 1 TO TBLEN
380 INDX1 = INDX1 + DELTA
390 TBL = SIN(INDX1)
400 HTBL$ = HEX$(TBL*16384)
410 RADS = INDX1 / PI
420 DEGR = NETDEG * (I - 1)
430 PRINT#3, "        DATA ";
440 PRINT#3, USING ">\  \";HTBL$;
450 PRINT#3, "         ANGLE = ";
460 PRINT#3, USING "###.####";DEGR;
470 PRINT#3, "  SINE = ";
480 PRINT#3, USING "#.######";TBL
490 NEXT
500 PRINT#3,"**********************************************************"
510 PRINT#3, "        PAGE"
520 CLOSE
530 PRINT "PROGRAM FINISHED"
540 END
```

```
**************************************************************
SINE    EQU  $           SINE TABLE LENGTH =  32
        DATA >0          ANGLE =   0.0000  SINE = 0.000000
        DATA >C7C        ANGLE =  11.2500  SINE = 0.195090
        DATA >187E       ANGLE =  22.5000  SINE = 0.382683
        DATA >238E       ANGLE =  33.7500  SINE = 0.555570
        DATA >2D41       ANGLE =  45.0000  SINE = 0.707107
        DATA >3537       ANGLE =  56.2500  SINE = 0.831470
        DATA >3B21       ANGLE =  67.5000  SINE = 0.923880
        DATA >3EC5       ANGLE =  78.7500  SINE = 0.980785
        DATA >4000       ANGLE =  90.0000  SINE = 1.000000
        DATA >3EC5       ANGLE = 101.2500  SINE = 0.980785
        DATA >3B21       ANGLE = 112.5000  SINE = 0.923880
        DATA >3537       ANGLE = 123.7500  SINE = 0.831470
        DATA >2D41       ANGLE = 135.0000  SINE = 0.707107
        DATA >238E       ANGLE = 146.2500  SINE = 0.555570
        DATA >187E       ANGLE = 157.5000  SINE = 0.382683
        DATA >C7C        ANGLE = 168.7500  SINE = 0.195090
        DATA >0          ANGLE = 180.0000  SINE = -.000000
        DATA >F384       ANGLE = 191.2500  SINE = -.195091
        DATA >E782       ANGLE = 202.5000  SINE = -.382684
        DATA >DC72       ANGLE = 213.7500  SINE = -.555571
        DATA >D2BF       ANGLE = 225.0000  SINE = -.707107
        DATA >CAC9       ANGLE = 236.2500  SINE = -.831470
        DATA >C4DF       ANGLE = 247.5000  SINE = -.923880
        DATA >C13B       ANGLE = 258.7500  SINE = -.980786
        DATA >C000       ANGLE = 270.0000  SINE = %-1.000000
        DATA >C13B       ANGLE = 281.2500  SINE = -.980785
        DATA >C4DF       ANGLE = 292.5000  SINE = -.923879
        DATA >CAC9       ANGLE = 303.7500  SINE = -.831469
        DATA >D2BF       ANGLE = 315.0000  SINE = -.707106
        DATA >DC72       ANGLE = 326.2500  SINE = -.555569
        DATA >E782       ANGLE = 337.5000  SINE = -.382682
        DATA >F384       ANGLE = 348.7500  SINE = -.195089
**************************************************************
        PAGE
```

Appendix D

TMS32010 Source Code

```
*
*************************************************************
***                  - DSP MODEM PROGRAM -                ***
*************************************************************
*** THIS CODE IMPLEMENTS A BELL 212A / V.22 MODEM         ***
*** ON THE TMS32010.                                      ***
***                                                       ***
*** SCRAMBLING AND DESCRAMBLING  ARE IMPLEMENTED          ***
*** ON THE TMS7742.                                       ***
*************************************************************
        IDT     'TASK6212'
        OPTION  XREF
        AORG    0
        B       START
********-------------------------------------------********
*****       DATA MEMORY USED.                         *****
********-------------------------------------------********
XDELTA EQU  0               * SWAVE MACRO CARRIER RATE
XALPHA EQU  1               * SWAVE MACRO CARRIER ANGLE
SINA   EQU  2               * XMIT SIN CARRIER MAGNETUDE
COSA   EQU  3               * XMIT COS CARRIER MAGNETUDE
ONE    EQU  4               * VALUE 1 HELD FOR MASKING
MASK1  EQU  5               * SWAVE MACRO TBL RANGE ADJ >7F
MASK2  EQU  6               * SWAVE MACRO TBL RANGE ADJ >7FFF
MASK3  EQU  7               * XMIT PHASE ENCODE MASK >0006
OFSET0 EQU  8               * SWAVE MACRO POINT TO COS TABLE
OFSET1 EQU  9               * XMIT POINT TO DIBIT ENCODE TABLE
XPTR   EQU  10              * XMIT POINT TO RAISED COS TABLE
CX0    EQU  11              * XMIT COEF FOR RAISED COS
CX1    EQU  12              * XMIT COEF FOR RAISED COS
CX2    EQU  13              * XMIT COEF FOR RAISED COS
XIBUF0 EQU  14              * XMIT STORE DATA FOR RAISED COS
XIBUF1 EQU  15              * XMIT STORE DATA FOR RAISED COS
XIBUF2 EQU  16              * XMIT STORE DATA FOR RAISED COS
XQBUF0 EQU  17              * XMIT STORE DATA FOR RAISED COS
XQBUF1 EQU  18              * XMIT STORE DATA FOR RAISED COS
XQBUF2 EQU  19              * XMIT STORE DATA FOR RAISED COS
XIOUT  EQU  20              * XMIT HOLD FILTERED I VALUE
XQOUT  EQU  21              * XMIT HOLD FILTERED Q VALUE
XMTOUT EQU  22              * XMIT HOLD FOR TRANSMIT OUTPUT
XOLDPH EQU  23              * XMIT HOLD LAST PHASE
XNEWPH EQU  24              * XMIT HOLD NEW PHASE
RDIBIT EQU  25              * DECODED DIBIT
INDXPH EQU  26              * XMIT POINT TO PHASE ENCODE TABLE
XDIBIT EQU  27              * XMIT DIBIT ISOLATION MASK
PLUS1  EQU  28              * +1 Q12 >FFF & MASK VALUE
XMTD   EQU  29              * XMIT HOLD DTE INPUT
RBUF0  EQU  30              * HOLD LOWPASS FILTERED SAMPLE
RBUF1  EQU  31              % RECEIVE BPF  COEFFICIENT
RBUF2  EQU  32              % RECEIVE BPF  COEFFICIENT
RBUF3  EQU  33              % RECEIVE BPF  COEFFICIENT
RBUF4  EQU  34              % RECEIVE BPF  COEFFICIENT
RBUF5  EQU  35              % RECEIVE BPF  COEFFICIENT
RBUF6  EQU  36              % RECEIVE BPF  COEFFICIENT
RBUF7  EQU  37              % RECEIVE BPF  COEFFICIENT
RBUF8  EQU  38              % RECEIVE BPF  COEFFICIENT
RBUF9  EQU  39              % RECEIVE BPF  COEFFICIENT
RBUF10 EQU  40              % RECEIVE BPF  COEFFICIENT
RBUF11 EQU  41              % RECEIVE BPF  COEFFICIENT
RBUF12 EQU  42              % RECEIVE BPF  COEFFICIENT
RBUF13 EQU  43              % RECEIVE BPF  COEFFICIENT
RBUF14 EQU  44              % RECEIVE BPF  COEFFICIENT
RBUF15 EQU  45              % RECEIVE BPF  COEFFICIENT
RBUF16 EQU  46              % RECEIVE BPF  COEFFICIENT
RBUF17 EQU  47              % RECEIVE BPF  COEFFICIENT
RBUF18 EQU  48              % RECEIVE BPF  COEFFICIENT
RBUF19 EQU  49              % RECEIVE BPF  COEFFICIENT
RBUF20 EQU  50              % RECEIVE BPF  COEFFICIENT
RBUF21 EQU  51              % RECEIVE BPF  COEFFICIENT
```

```
RBUF22 EQU  52             % RECEIVE BPF  COEFFICIENT
RBUF23 EQU  53             % RECEIVE BPF  COEFFICIENT
RBUF24 EQU  54             % RECEIVE BPF  COEFFICIENT
RBUF25 EQU  55             % RECEIVE BPF  COEFFICIENT
RBUF26 EQU  56             % RECEIVE BPF  COEFFICIENT
RBUF27 EQU  57             % RECEIVE BPF  COEFFICIENT
RBUF28 EQU  58             % RECEIVE BPF  COEFFICIENT
RBUF29 EQU  59             % RECEIVE BPF  COEFFICIENT
RBUF30 EQU  60             % RECEIVE BPF  COEFFICIENT
RBUF31 EQU  61             % RECEIVE BPF  COEFFICIENT
RBUF32 EQU  62             % RECEIVE BPF  COEFFICIENT
RBUF33 EQU  63             % RECEIVE BPF  COEFFICIENT
RBUF34 EQU  64             % RECEIVE BPF  COEFFICIENT
RBUF35 EQU  65             % RECEIVE BPF  COEFFICIENT
RBUF36 EQU  66             % RECEIVE BPF  COEFFICIENT
RBUF37 EQU  67             % RECEIVE BPF  COEFFICIENT
AGC    EQU  68             * AUTOMATIC GAIN FACTOR
AGCRA  EQU  69             * SIGNAL MAX RUNNING AVERAGE FOR AGC
RECST  EQU  70             * RECEIVER STATUS
AGCOFF EQU  71             * AGC CALCULATION LOOKUP TABLE
BSMAX  EQU  72             * BAUD SIGNAL MAX
AGCNT  EQU  73             * BAUD SAMPLE COUNT
AGCLEV EQU  74             * TEMPORARY AGC LEVEL (AGCUPT)
SAMPLE EQU  75             * BAUD LIMIT SAMPLE COUNT
SAMXMT EQU  76             * TRANSMITTER SAMPLE COUNT
BITOUT EQU  77             * DIBIT POSITIONED TO XMIT TO 7041
RPHSE  EQU  78             * OFFSET FOR RECEIVE PHASE DECODE
TRSHD1 EQU  79             * THRESHOLD FOR CARRIER RECOVERY
RALPHA EQU  80             * RECEIVE CARRIER POINTER
RDELTA EQU  81             * DELTA TO GENERATE RECEIVE CARRIER
ISUM   EQU  82             * FILTERED/PHASE SHIFTED SAMPLE
QSUM   EQU  83             * FILTERED/PHASE SHIFTED SAMPLE
RECI   EQU  84             * BASEBAND I CHANNEL
ROLDPH EQU  85             * PREVIOUS ABSOLUTE PHASE (QUADRANT)
RNEWPH EQU  86             * CURRENT  ABSOLUTE PHASE (QUADRANT)
ERRSIG EQU  87             * FILTERED CARRIER ERROR SIGNAL
MINUS1 EQU  88             * MINUS 1 IN THE Q12 FORMAT
PLL1   EQU  89             * CARRIER RECOVERY PLL FILTER COEFFICIENT 1
PLL2   EQU  90             * CARRIER RECOVERY PLL FILTER COEFFICIENT 2
FOUR   EQU  91             * >4 ( MASK VALUE FOR PHASE CODE/DECODE)
SIGNI  EQU  92             * SIGN OF I CHANNEL (TO COMPUTE CARRIER ERROR)
SIGNQ  EQU  93             * SIGN OF Q CHANNEL (TO COMPUTE CARRIER ERROR)
ERROR  EQU  94             * CARRIER PHASE ERROR
TEMP   EQU  95             * MISC. TEMPERORY REGISTER
RECQ   EQU  96             * BASEBAND Q CHANNEL
*-----DEFINE REGISTERS FOR BAUD CLOCK
ENRGY  EQU  97             * CURRENT ENERGY
PENRGY EQU  98             * PREVIOUS ENERGY
BERROR EQU  99             * BAUD CLOCK ERROR
BEROUT EQU  100            * OUTPUT OF BAUD PLL LOOP FILTER
BPLL1  EQU  101            * CLOCK RECOVERY PLL FILTER COEFFICIENT 1
BPLL2  EQU  102            * CLOCK RECOVERY PLL FILTER COEFFICIENT 2
TRSHD2 EQU  103            * CLOCK RECOVERY TRESHOLD
*---------------
*****       TRANSMITTER DIBIT ENCODER TABLE.        *****
********-------------------------------------------********
ENCODE  DATA    >0002      * DIBIT '01' = 90 deg.
        DATA    >0000      * DIBIT '00' = 0 deg.
        DATA    >0004      * DIBIT '10' = 180 deg
        DATA    >0006      * DIBIT '11' = 270 deg.
XPHASE  DATA    >7FFF      * 0 deg.    I CHANNEL =  1
        DATA    >0000      *           Q CHANNEL =  0
        DATA    >0000      * 90 deg.   I CHANNEL =  0
        DATA    >8000      *           Q CHANNEL = -1
        DATA    >8000      * 180 deg.  I CHANNEL = -1
        DATA    >0000      *           Q CHANNEL =  0
        DATA    >0000      * 270 deg.  I CHANNEL =  0
        DATA    >7FFF      *           Q CHANNEL =  1
********-------------------------------------------********
```

```
*****         RECEIVER DIBIT ENCODER TABLE.                    *****
*****         DIBITS are formed as 'MSB,LSB'.                  *****
********---------------------------------------------------********
RPHASE  DATA     >0001        * 0  deg.,  DIBIT = '01'
        DATA     >0000        * 90 deg.,  DIBIT = '00'
        DATA     >0002        * 180 deg., DIBIT = '10'
        DATA     >0003        * 270 deg., DIBIT = '11'
*
M1      DATA     >7FFF        * MASK 1
M2      DATA     >007F        * MASK 2
M3      DATA     >0006        * MASK 3
CK      DATA     >0208        * CLOCK FOR AIB
MD      DATA     >000A        * MODE FOR AIB
ST      DATA     >1800
        DATA     >0000
DT      DATA     >1000        * TRANSMIT DELTA.
        DATA     >2000        * RECEIVE DELTA.
TH1     DATA     >0007        *  0.01  Q12 TRSHD FOR CARRIER
TH2     DATA     >0008        *  0.01  Q12 TRSHD FOR BAUD CLOCK
MIN1    DATA     >F000        * -1 Q12
PLS1    DATA     >0FFF        *  1 Q12
*
********---------------------------------------------------********
*****     PLL LOOP FILTER COFFECIENTS.                         *****
********---------------------------------------------------********
*
PLLC1   DATA     >4500    * Q15   CARRIER PLL INITIAL COEF. 1
PLLC2   DATA     >80      * Q15   CARRIER PLL COEFFICIENT 2
BPLLC1  DATA     >4000    * Q15   BAUD CLOCK PLL INITIAL COEF. 1
BPLLC2  DATA     >50      * Q15   BAUD CLOCK PLL COEFFICIENT 2
PLLC    DATA     >7A00    * Q15   CARRIER PLL STEADY STATE COEF. 1
BPLLC   DATA     >7500    * Q15   BAUD CLOCK PLL STEADY STATE COEF. 1
*
**********-------------------------------------------------********
*****     TASK MASTER SEQUENCE TABLE  (RECEIVE)                *****
*****     TASKS ARE EXECUTED FROM BOTTOM TO TOP                *****
**********-------------------------------------------------********
*
TSKSEQ  EQU  $
        DATA DUMMY    UNUSED CYCLE                             15
        DATA DUMMY    UNUSED CYCLE                             14
        DATA DUMMY    UNUSED CYCLE                             13
        DATA DUMMY    UNUSED CYCLE                             12
        DATA BDCLK2   COMPUTE ENERGY E(11)                     11
        DATA DUMMY    UNUSED CYCLE                             10
        DATA OUT      COMMUNICATE WITH TMS7742                  9
        DATA DECODE   DECODE/GET SCRAMBLED DIBIT                8
        DATA DEMODB   DEMODULATE IN THE MIDDLE OF BAUD          7
        DATA DUMMY    UNUSED CYCLE                              6
        DATA AGCUPT   UPDATE THE AGC (IF NECESSARY)             5
        DATA DUMMY    UNUSED CYCLE                              4
        DATA BDCLK1   COMPUTE ENERGY E(3)                       3
        DATA DUMMY    UNUSED CYCLE                              2
        DATA DUMMY    UNUSED CYCLE                              1
        DATA DUMMY    UNUSED CYCLE                              0
*
**********-------------------------------------------------********
*****     TASK MASTER SEQUENCE TABLE  (TRANSMIT)               *****
*****     TASKS ARE EXECUTED FROM BOTTOM TO TOP                *****
**********-------------------------------------------------********
*
TSKXMT  EQU  $
        DATA GETDBT   GET THE NEXT DIBIT                       16
        DATA DUMXMT   NO CYCLE                                 15
        DATA DUMXMT   NO CYCLE                                 14
        DATA DUMXMT   NO CYCLE                                 13
        DATA DUMXMT   NO CYCLE                                 12
        DATA DUMXMT   NO CYCLE                                 11
        DATA DUMXMT   NO CYCLE                                 10
```

```
        DATA DUMXMT   NO CYCLE                                   9
        DATA DUMXMT   NO CYCLE                                   8
        DATA DUMXMT   NO CYCLE                                   7
        DATA DUMXMT   NO CYCLE                                   6
        DATA DUMXMT   NO CYCLE                                   5
        DATA DUMXMT   NO CYCLE                                   4
        DATA DUMXMT   NO CYCLE                                   3
        DATA DUMXMT   NO CYCLE                                   2
        DATA DUMXMT   NO CYCLE                                   1
        PAGE
********-----------------------------------------------********
*****      RAISED COSINE COEFFICIENT TABLE.                *****
********-----------------------------------------------********
COEF    DATA     >1
        DATA     >49A
        DATA     >394
        DATA     >FFD9
        DATA     >5A2
        DATA     >29A
        DATA     >FFAB
        DATA     >6A0
        DATA     >1B5
        DATA     >FF7A
        DATA     >789
        DATA     >ED
        DATA     >FF4C
        DATA     >853
        DATA     >45
        DATA     >FF27
        DATA     >8F4
        DATA     >FFC3
        DATA     >FF11
        DATA     >963
        DATA     >FF65
        DATA     >FF10
        DATA     >99C
        DATA     >FF2A
        DATA     >FF2A
        DATA     >99C
        DATA     >FF10
        DATA     >FF65
        DATA     >963
        DATA     >FF11
        DATA     >FFC3
        DATA     >8F4
        DATA     >FF27
        DATA     >45
        DATA     >853
        DATA     >FF4C
        DATA     >ED
        DATA     >789
        DATA     >FF7A
        DATA     >1B5
        DATA     >6A0
        DATA     >FFAB
        DATA     >29A
        DATA     >5A2
        DATA     >FFD9
        DATA     >394
        DATA     >49A
        DATA     >1
*
***************************************************************
*****   AGC DIVIDE LOOKUP TABLE                          *****
*****     STANDARD GAIN RANGE -- >3CC3 - >3F79           *****
*****     WITH 5% SIGNAL VARIATION -- >3966 - >41D6      *****
*****                                                    *****
***************************************************************
```

```
AGCTBL EQU  $-32             AGC TABLE LENGTH =  32
       DATA >F8                        0.9687500
       DATA >F0                        0.9375000
       DATA >EA                        0.9140625
       DATA >E3                        0.8867188 35 -
       DATA >DD                        0.8632812
       DATA >D7                        0.8398438
       DATA >D2                        0.8203125
       DATA >CC                        0.7968750 39 -
       DATA >C7                        0.7773438
       DATA >C3                        0.7617188
       DATA >BE                        0.7421875
       DATA >BA                        0.7265625 43 -
       DATA >B6                        0.7109375
       DATA >B2                        0.6953125
       DATA >AE                        0.6796875
       DATA >AA                        0.6640625 47 -
       DATA >A7                        0.6523438
       DATA >A3                        0.6367188
       DATA >A0                        0.6250000
       DATA >9D                        0.6132813 51 -
       DATA >9A                        0.6015625
       DATA >97                        0.5898438
       DATA >94                        0.5781250
       DATA >92                        0.5703125 55 -
       DATA >8F                        0.5585938
       DATA >8D                        0.5507813
       DATA >8A                        0.5390625
       DATA >88                        0.5312500 59 -
       DATA >86                        0.5234375
       DATA >84                        0.5156250
       DATA >82                        0.5078125
       DATA >7F                        0.4960938 63 -
       PAGE
*
********------------------------------------------------********
*****             SINE(COSINE) TABLE                       *****
********------------------------------------------------********
SINE    DATA      >0
        DATA      >648
        DATA      >C8C
        DATA      >12C8
        DATA      >18F9
        DATA      >1F1A
        DATA      >2528
        DATA      >2B1F
        DATA      >30FC
        DATA      >36BA
        DATA      >3C57
        DATA      >41CE
        DATA      >471D
        DATA      >4C40
        DATA      >5134
        DATA      >55F6
        DATA      >5A82
        DATA      >5ED7
        DATA      >62F2
        DATA      >66D0
        DATA      >6A6E
        DATA      >6DCA
        DATA      >70E3
        DATA      >73B6
        DATA      >7642
        DATA      >7885
        DATA      >7A7D
        DATA      >7C2A
        DATA      >7D8A
        DATA      >7E9D
        DATA      >7F62
```

```
        DATA    >7FD9
COSINE  DATA    >7FFF
        DATA    >7FD9
        DATA    >7F62
        DATA    >7E9D
        DATA    >7D8A
        DATA    >7C2A
        DATA    >7A7D
        DATA    >7885
        DATA    >7642
        DATA    >73B6
        DATA    >70E3
        DATA    >6DCA
        DATA    >6A6E
        DATA    >66D0
        DATA    >62F2
        DATA    >5ED7
        DATA    >5A82
        DATA    >55F6
        DATA    >5134
        DATA    >4C40
        DATA    >471D
        DATA    >41CE
        DATA    >3C57
        DATA    >36BA
        DATA    >30FC
        DATA    >2B1F
        DATA    >2528
        DATA    >1F1A
        DATA    >18F9
        DATA    >12C8
        DATA    >C8C
        DATA    >648
        DATA    >0
        DATA    >F9B8
        DATA    >F374
        DATA    >ED38
        DATA    >E707
        DATA    >E0E6
        DATA    >DAD8
        DATA    >D4E1
        DATA    >CF04
        DATA    >C946
        DATA    >C3A9
        DATA    >BE32
        DATA    >B8E3
        DATA    >B3C0
        DATA    >AECC
        DATA    >AA0A
        DATA    >A57E
        DATA    >A129
        DATA    >9D0E
        DATA    >9930
        DATA    >9592
        DATA    >9236
        DATA    >8F1D
        DATA    >8C4A
        DATA    >89BE
        DATA    >877B
        DATA    >8583
        DATA    >83D6
        DATA    >8276
        DATA    >8163
        DATA    >809E
        DATA    >8027
        DATA    >8000
        DATA    >8027
        DATA    >809E
        DATA    >8163
```

```
          DATA      >8276
          DATA      >83D6
          DATA      >8583
          DATA      >877B
          DATA      >89BE
          DATA      >8C4A
          DATA      >8F1D
          DATA      >9236
          DATA      >9592
          DATA      >9930
          DATA      >9D0E
          DATA      >A129
          DATA      >A57E
          DATA      >AA0A
          DATA      >AECC
          DATA      >B3C0
          DATA      >B8E3
          DATA      >BE32
          DATA      >C3A9
          DATA      >C946
          DATA      >CF04
          DATA      >D4E1
          DATA      >DAD8
          DATA      >E0E6
          DATA      >E707
          DATA      >ED38
          DATA      >F374
          DATA      >F9B8
*
********-------------------------------------------------------********
*****  RECEIVER I CHANNEL BPASS FILTER COEFFICIENTS   *****
********-------------------------------------------------------********
*
ICF0      EQU      58       * 3A 0.014064
*ICF1     EQU      0        * 0 0.000000
ICF2      EQU      -58      * FFC6 -0.014067
*ICF3     EQU      0        * 0 0.000000
ICF4      EQU      28       * 1C 0.006883
*ICF5     EQU      0        * 0 0.000000
ICF6      EQU      37       * 25 0.009069
*ICF7     EQU      0        * 0 0.000000
ICF8      EQU      -137     * FF77 -0.033477
*ICF9     EQU      0        * 0 0.000000
ICF10     EQU      262      * 106 0.063862
*ICF11    EQU      0        * 0 0.000000
ICF12     EQU      -393     * FE77 -0.095882
*ICF13    EQU      0        * 0 0.000000
ICF14     EQU      509      * 1FD 0.124198
*ICF15    EQU      0        * 0 0.000000
ICF16     EQU      -588     * FDB4 -0.143676
*ICF17    EQU      0        * 0 0.000000
ICF18     EQU      617      * 269 0.150616
*ICF19    EQU      0        * 0 0.000000
ICF20     EQU      -588     * FDB4 -0.143676
*ICF21    EQU      0        * 0 0.000000
ICF22     EQU      509      * 1FD 0.124198
*ICF23    EQU      0        * 0 0.000000
ICF24     EQU      -393     * FE77 -0.095882
*ICF25    EQU      0        * 0 0.000000
ICF26     EQU      262      * 106 0.063862
*ICF27    EQU      0        * 0 0.000000
ICF28     EQU      -137     * FF77 -0.033477
*ICF29    EQU      0        * 0 0.000000
ICF30     EQU      37       * 25 0.009069
*ICF31    EQU      0        * 0 0.000000
ICF32     EQU      28       * 1C 0.006883
*ICF33    EQU      0        * 0 0.000000
ICF34     EQU      -58      * FFC6 -0.014067
*ICF35    EQU      0        * 0 0.000000
ICF36     EQU      58       * 3A 0.014064
```

```
*
********------------------------------------------------------********
*****  RECEIVER Q CHANNEL BPASS FILTER COEFFICIENTS  *****
********------------------------------------------------------********
*
*QCF0    EQU     0       * 0 0.000000
QCF1     EQU     61      * 3D 0.014809
*QCF2    EQU     0       * 0 0.000000
QCF3     EQU     -47     * FFD1 -0.011510
*QCF4    EQU     0       * 0 0.000000
QCF5     EQU     0       * 0 0.000034
*QCF6    EQU     0       * 0 0.000000
QCF7     EQU     83      * 53 0.020321
*QCF8    EQU     0       * 0 0.000000
QCF9     EQU     -197    * FF3B -0.048158
*QCF10   EQU     0       * 0 0.000000
QCF11    EQU     328     * 148 0.079991
*QCF12   EQU     0       * 0 0.000000
QCF13    EQU     -454    * FE3A -0.110844
*QCF14   EQU     0       * 0 0.000000
QCF15    EQU     554     * 22A 0.135320
*QCF16   EQU     0       * 0 0.000000
QCF17    EQU     -610    * FD9E -0.148859
*QCF18   EQU     0       * 0 0.000000
QCF19    EQU     610     * 262 0.148859
*QCF20   EQU     0       * 0 0.000000
QCF21    EQU     -554    * FDD6 -0.135320
*QCF22   EQU     0       * 0 0.000000
QCF23    EQU     454     * 1C6 0.110844
*QCF24   EQU     0       * 0 0.000000
QCF25    EQU     -328    * FEB8 -0.079991
*QCF26   EQU     0       * 0 0.000000
QCF27    EQU     197     * C5 0.048158
*QCF28   EQU     0       * 0 0.000000
QCF29    EQU     -83     * FFAD -0.020321
*QCF30   EQU     0       * 0 0.000000
QCF31    EQU     0       * 0 -0.000034
*QCF32   EQU     0       * 0 0.000000
QCF33    EQU     47      * 2F 0.011510
*QCF34   EQU     0       * 0 0.000000
QCF35    EQU     -61     * FFC3 -0.014809
*QCF36   EQU     0       * 0 0.000000
*
         PAGE
*************************************************************
******     Initializtion routine                      ******
*************************************************************
START    DINT
         LDPK    0
         ROVM
         LACK    1
         SACL    ONE
         LACK    M1          * INITIALIZE MASK 1
         TBLR    MASK1
         LACK    M2          * INITIALIZE MASK 2
         TBLR    MASK2
         LACK    M3          * INITIALIZE MASK 3
         TBLR    MASK3
*
         LACK    MD              * AIB BOARD INITIALIZATION.
         TBLR    TEMP            * MD IS MODE CNTRL FOR AIB.
         OUT     TEMP,PA0
         LACK    CK              * CK IS SAMPLE RATE FOR AIB.
         TBLR    TEMP
         OUT     TEMP,PA1        * SENT CLOCK VALUE TO PORT 1(NEW AIB)
*
         LACK    SINE            * TABLE OFFSET INITIALIZATION.
         SACL    OFSETO          * SINE TABLE OFFSET
         LACK    ENCODE
```

```
         SACL     OFSET1         * <DIBIT TO PHASE> TABLE
         LACK     COEF
         SACL     XPTR           * RAISED COS COEF. TABLE.
         LACK     XPHASE
         SACL     INDXPH         * OFSET FOR XMIT PHASE TABLE
         LACK     RPHASE
         SACL     RPHSE          * OFSET FOR RCVR PHASE TABLE
         LACK     4
         SACL     FOUR           *
         ZAC                     * MISC. INITIALIZATIONS.
         SACL     ROLDPH         * INITIALIZE PREVIOUS TOTAL PHASE
*
         SACL     RALPHA         * SWAVE INITIALIZATIONS.
         SACL     XALPHA
         LACK     DT
         TBLR     XDELTA         * READ SWAVE DELTAs
         ADD      ONE
         TBLR     RDELTA
*
*
         LACK     TH1            * CARRIER PLL THRESHOLD
         TBLR     TRSHD1
         ADD      ONE
         TBLR     TRSHD2         * BAUD CLOCK PLL THRESHOLD
         LACK     MIN1
         TBLR     MINUS1         * -1 IN Q12
         LACK     PLS1
         TBLR     PLUS1          * +1 IN Q12
*-----------------------------------
         LACK     PLLC1          * CARRIER PLL INITIAL COEF. 1
         TBLR     PLL1
         ADD      ONE
         TBLR     PLL2           * CARRIER PLL COEF. 2
         ADD      ONE
         TBLR     BPLL1          * BAUD CLOCK PLL INITIAL COEF. 1
         ADD      ONE
         TBLR     BPLL2          * BAUD CLOCK COEF. 2
*-----------------------------------
         LACK     AGCTBL         * SET THE AGC TABLE LOOKUP
         SACL     AGCOFF         * OFF SET VALUE
         LAC      ONE,13         * INITIALIZE RUNNING AVERAGE
         SACL     AGCRA          * TO >2000
         LACK     >FF            * INITIALIZE THE AGC FACTOR
         SACL     AGC            * TO ONE
         ZAC                     * INITIALIZE THE
         SACL     BSMAX          * BAUD SIGNAL MAX TO ZERO
         LACK     3              * RUNNING AVERAGE COUNT
         SACL     AGCNT          * TO THREE
         LACK     >20            * SET THE ENERGY DETECT
         SACL     RECST          * BIT IN THE STATUS FLAG WORD
*
         LACK     15             * SET THE REC SAMPLE COUNT
         SACL     SAMPLE         * TO 16
         ZAC                     * SET THE XMT SAMPLE COUNT
         SACL     SAMXMT         * TO ZERO
*
         PAGE
*-------------------------------------------------------------------*
*   THE FOLLOWING CODE HANDLES COMMANDS FROM THE 7042               *
*-------------------------------------------------------------------*
        LAC   ONE,4        SET COUNTER VALUE TO RUN
        SACL  TEMP         DLB AT 600 BAUD
COMD    BIOZ  LOOK          WAIT FOR 9600HZ SAMPLE PULSE
        B     COMD
LOOK    NOP
        IN    RBUF0,PA2    *--- DUMMY READ TO GET COUNTER GOING
LOOK1   IN    XMTD,PA6     LOOK FOR COMMAND
        LACK  >30          MASK OFF ALL BUT COMMAND BITS
        AND   XMTD         CHECK COMMAND BITS FOR NEW COMMAND
```

```
        BZ    COMD              IF ZERO THEN NO COMMAND YET
        SUB   ONE,4             CHECK FOR DIGITAL LOOP BACK TEST
        BZ    LDLB              IF SO THEN EXECUTE TEST
        SUB   ONE,4             CHECK FOR MODEM RUN COMMAND
        BZ    WAIT              IF SO THEN RUN MODEM
        LACK  >F                MASK OFF COMMAND BITS
        AND   XMTD              TO GET SPECIFIC CONFIGURATION
*   THIS IS FOR CONFIGURATION CODES
        BZ    COMD              ZERO IS NOT VALID COMMAND
        SUB   ONE               CHECK FOR COMMAND ONE
        BZ    SETALB            SETUP THE MODEM TO RUN ALB
        SUB   ONE               CHECK FOR COMMAND TWO
        BZ    SETORG            SETUP THE MODEM TO RUN ORIGINATE
        SUB   ONE               CHECK FOR COMMAND THREE
        BZ    SQTREC            SHUT DOWN RECEIVER TO RUN XMIT ONLY
        B     COMD              CHECK FOR NEXT COMMAND
SETALB  LAC   ONE,13            LOAD ACC WITH 2000 TO PUT
        SACL  XDELTA            XMIT IN SAME BAND AS RECEIVE
        B     COMD              CHECK FOR NEXT COMMAND
SETORG  LAC   ONE,12            LOAD ACC WITH 1000 TO PUT
        SACL  XDELTA            XMIT IN ORIGINATE MODE
        B     COMD              CHECK FOR NEXT COMMAND
SQTREC  LAC   ONE,8             SET RECEIVER SQUELCH BIT
        OR    RECST             IN THE RECEIVE STATUS REG
        SACL  RECST             TO DISABLE RECEIVER CODE
        B     COMD              CHECK FOR NEXT COMMAND
*
LDLB    BIOZ  DLBOUT            WAIT FOR NEXT SAMPLE PERIOD
        B     LDLB              LOOP ON TIMER
DLBOUT  IN    RBUF0,PA2         *--- DUMMY READ TO GET COUNTER GOING
        LAC   TEMP              GET 16 SAMPLE BAUD COUNTER
        SUB   ONE               DECREMENT IT
        SACL  TEMP              SAVE COUNT
        BNZ   LDLB              COUNT ANOTHER SAMPLE PERIOD
        LAC   ONE,4             RESET COUNTER VALUE TO RUN
        SACL  TEMP              DLB AT 600 BAUD
        LAC   XMTD,10           ADJUST FOR OUTPUT RANGE
        OR    MINUS1          * MASK COMMAND BITS(15-12)TO 1'S
        SACL  XMTD              STORE IT FOR OUTPUT
        OUT   XMTD,PA6          ECHO INPUT
        B     LOOK1             REPEAT LOOP BACK TEST
*-------------------------------------------------------------------*
********--------------------------------------------------********
*****    THE FOLLOWING SECTION IMPLEMENTS MODEM FUNCTIONS    *****
********--------------------------------------------------********
*-------------------------------------------------------------------*
WAIT     BIOZ      GO             * WAIT FOR 9600HZ SAMPLE PULSE
         B         WAIT
GO       NOP
         OUT       XMTOUT,PA2     * OUTPUT TO D/A
         IN        RBUF0,PA2      * INPUT FROM A/D
********-------------------------------------------------********
*****    TRANSMITER SECTION STARTS HERE.                    *****
********-------------------------------------------------********
XMITER   EQU       $
*
********-------------------------------------------------********
*****       SINE(COSINE) WAVE GENERATION                    *****
********-------------------------------------------------********
SWAVE    EQU       $
         LAC       XALPHA,8           * DELTA IS THE INCREMENT.
         SACH      TEMP               * ISOLATE INTEGER PORTION.
         LAC       TEMP
         ADD       OFSETO             * ADD INDEX TO SINE TABLE.
         TBLR      SINA               * SINE VALUE,  (Q15).
         LACK      >20                * OFFSET TO COSINE VALUE (Q15).
         ADD       TEMP
         AND       MASK2
         ADD       OFSETO             * ADD INDEX TO COSINE TABLE.
```

```
        TBLR    COSA                   * COSINE VALUE, (Q15).
        LAC     XALPHA                 * COMPUTE ADDRESS OF NEXT
        ADD     XDELTA                 * POINT FOR TABLE.
        AND     MASK1                  * KEEP MOD128, MASK=>7FFF.
        SACL    XALPHA                 * SAVE NEXT ADDRESS
*
********-----------------------------------------------------********
*****   TRANSMITTER 48 TAP RAISED COSINE FILTER.                *****
*****   INPUTS UPDATED AT 600HZ RATE.                           *****
*****   OUTPUT UPDATED AT 9600HZ RATE.                          *****
********-----------------------------------------------------********
RACS    EQU     $
        LAC     XPTR
        TBLR    CX0             * RETRIEVE COEFFICIENTS
        ADD     ONE
        TBLR    CX1
        ADD     ONE
        TBLR    CX2
        ADD     ONE
        SACL    XPTR
*
        ZAC
        LT      XIBUF2          * COMPUTE FILTER TAPS ICHAN.
        MPY     CX2
        LTA     XIBUF1
        MPY     CX1
        LTA     XIBUF0
        MPY     CX0
        APAC
        SACH    XIOUT,1
*
        ZAC
        LT      XQBUF2          * COMPUTE FILTER TAPS QCHAN.
        MPY     CX2
        LTA     XQBUF1
        MPY     CX1
        LTA     XQBUF0
        MPY     CX0
        APAC
        SACH    XQOUT,1
*
XMIT    ZAC
        LT      XIOUT           * ICHAN*cos(wt)+ QCHAN*sin(wt)
        MPY     COSA
        LTA     XQOUT
        MPY     SINA
        APAC
        SACH    XMTOUT,1
*
        PAGE
********-----------------------------------------------------********
*****   RECEIVER I CHANNEL BANDPASS FILTER.                     *****
*****   SAMPLING RATE IS 9600HZ.                                *****
********-----------------------------------------------------********
CONT6   ZAC
        LT      RBUF36
        MPYK    ICF36
        LTA     RBUF34
        MPYK    ICF34
        LTA     RBUF32
        MPYK    ICF32
        LTA     RBUF30
        MPYK    ICF30
        LTA     RBUF28
        MPYK    ICF28
        LTA     RBUF26
        MPYK    ICF26
        LTA     RBUF24
```

```
        MPYK    ICF24
        LTA     RBUF22
        MPYK    ICF22
        LTA     RBUF20
        MPYK    ICF20
        LTA     RBUF18
        MPYK    ICF18
        LTA     RBUF16
        MPYK    ICF16
        LTA     RBUF14
        MPYK    ICF14
        LTA     RBUF12
        MPYK    ICF12
        LTA     RBUF10
        MPYK    ICF10
        LTA     RBUF8
        MPYK    ICF8
        LTA     RBUF6
        MPYK    ICF6
        LTA     RBUF4
        MPYK    ICF4
        LTA     RBUF2
        MPYK    ICF2
        LTA     RBUF0
        MPYK    ICF0
        APAC
        SACH    ISUM,4        * OUTPUT OF I CHAN.
*
********_________________________________________********
*****   RECEIVER Q CHANNEL BANDPASS FILTER.         *****
*****   SAMPLING RATE IS 9600HZ.                    *****
********_________________________________________********
        LTD     RBUF35
        ZAC
        MPYK    QCF35
        DMOV    RBUF34
        LTD     RBUF33
        MPYK    QCF33
        DMOV    RBUF32
        LTD     RBUF31
        MPYK    QCF31
        DMOV    RBUF30
        LTD     RBUF29
        MPYK    QCF29
        DMOV    RBUF28
        LTD     RBUF27
        MPYK    QCF27
        DMOV    RBUF26
        LTD     RBUF25
        MPYK    QCF25
        DMOV    RBUF24
        LTD     RBUF23
        MPYK    QCF23
        DMOV    RBUF22
        LTD     RBUF21
        MPYK    QCF21
        DMOV    RBUF20
        LTD     RBUF19
        MPYK    QCF19
        DMOV    RBUF18
        LTD     RBUF17
        MPYK    QCF17
        DMOV    RBUF16
        LTD     RBUF15
        MPYK    QCF15
        DMOV    RBUF14
        LTD     RBUF13
        MPYK    QCF13
        DMOV    RBUF12
```

```
         LTD       RBUF11
         MPYK      QCF11
         DMOV      RBUF10
         LTD       RBUF9
         MPYK      QCF9
         DMOV      RBUF8
         LTD       RBUF7
         MPYK      QCF7
         DMOV      RBUF6
         LTD       RBUF5
         MPYK      QCF5
         DMOV      RBUF4
         LTD       RBUF3
         MPYK      QCF3
         DMOV      RBUF2
         LTD       RBUF1
         MPYK      QCF1
         DMOV      RBUF0
         APAC
         SACH      QSUM,4         * OUTPUT OF Q CHAN.
*
         PAGE
**********************************************************************
*        DETECT MAXIMUM SIGNAL STRENGTH OF RECI PER BAUD
AGCAL    EQU       $
         LAC       ISUM           * AGC VALUE CALCULATED USING ISUM
         ABS                        GET MAGNETUDE OF SIGNAL
         SUB       BSMAX            COMPARE TO PREVIOUS MAX VALUE
         BLZ       OVRMAX           IF LESS THAN THEN JUMP OVER UPDATE
         ADD       BSMAX            RESTORE VALUE AND
         SACL      BSMAX            STORE AS NEW MAX
*        MULTIPLY IN AGC FACTOR TO FILTERED SIGNAL
OVRMAX   LT        AGC              MULTIPLY THE AGC FACTOR
MPYI     MPY       ISUM             BY THE FILTERED DATA ELEMENT
         PAC                        MOVE THE PRODUCT TO THE ACC
         SACH      TEMP,4           SAVE TOP HALF OF ACC
         AND       PLUS1            MASK OFF UNUSABLE BITS
         SACL      ISUM             SAVE BOTTOM HALF OF ACC
         ZALH      TEMP             RELOAD HIGH ACC VALUE
         ADD       ISUM,4         SHIFT LOW HALF INTO POSITION
         SACH      ISUM,4         STORE Q15 GAINED FILTERED DATA
MPYQ     LT        AGC              MULTIPLY THE AGC FACTOR
         MPY       QSUM             BY THE FILTERED DATA ELEMENT
         PAC                        MOVE THE PRODUCT TO THE ACC
         SACH      TEMP,4           SAVE TOP HALF OF ACC
         AND       PLUS1            MASK OFF UNUSABLE BITS
         SACL      QSUM             SAVE BOTTOM HALF OF ACC
         ZALH      TEMP             RELOAD HIGH ACC VALUE
         ADD       QSUM,4         SHIFT LOW HALF INTO POSITION
         SACH      QSUM,4         STORE Q15 GAINED FILTERED DATA
*
         PAGE
*****************************************************************
*  The following code is the time sliced code task master.      *
*  The routine monitors the status of the modem operations      *
*  and sequences the code appropriately.                        *
*****************************************************************
MASTER   EQU  $
         LAC  ONE,5        *  CHECK OPERATING STATUS FOR
         AND  RECST        *  ENERGY DETECT
         BZ   HANGUP       *  IF NO ENERGY DETECT THEN HANG UP
         LAC  ONE,4        *  CHECK IF LOCAL CARRIER
         AND  RECST        *  IS LOCKED. IF SO SWITCH
         BNZ  CARLCK       *  PLL FILTERS BANDWIDTH
         B    NORMAL       *  EXECUTE NORMAL SEQUENCE
*
CARLCK   LACK PLLC         * CHANGE CARRIER PLL COEF. 1
         TBLR PLL1
         LACK BPLLC        * CHANGE BAUD CLOCK PLL COEF. 1
```

```
        TBLR BPLL1
NORMAL  EQU  $
        LAC  SAMPLE         * DECREMENT THE SAMPLE COUNT
        SUB  ONE            * TO CHECK FOR END OF BAUD
        BGEZ OVRSAM         * IF NOT THEN SKIP COUNT RESET
        LACK 15             * RESTART THE SAMPLE COUNTER AT 15
OVRSAM  SACL SAMPLE         * SAVE NEW COUNT VALUE
        LACK TSKSEQ         * GET ADDRESS OF TOP OF TABLE
        ADD  SAMPLE         * ADD IN OFFSET
        TBLR TEMP           * GET THE PROGRAM ADDRESS
        LAC  TEMP           * FOR THE TASK CALL
        CALA                * EXECUTE THE APPROPRIATE TASK
*
*    UPDATE CARRIER ANGLE AT SAMPLE RATE
*
        LAC  RALPHA         * COMPUTE ADDRESS OF NEXT
        ADD  RDELTA         * POINT FOR TABLE.
        AND  MASK1          * KEEP MOD128, MASK=>7FFF.
        SACL RALPHA         * SAVE NEXT ADDRESS
*
MASXMT  EQU  $              * EXECUTE TRANSMIT TASK SEQUENCE
        LAC  SAMXMT         * DECREMENT THE SAMPLE COUNT
        SUB  ONE            * TO CHECK FOR END OF BAUD
        BGEZ OVRSM1         * IF NOT THEN SKIP COUNT RESET
        LACK 15             * RESTART THE SAMPLE COUNTER AT 15
OVRSM1  SACL SAMXMT         * SAVE NEW COUNT VALUE
        LACK TSKXMT         * GET ADDRESS OF TOP OF TABLE
        ADD  SAMXMT         * ADD IN OFFSET
        TBLR TEMP           * GET THE PROGRAM ADDRESS
        LAC  TEMP           * FOR THE TASK CALL
        CALA                * EXECUTE THE APPROPRIATE TASK
        B    WAIT           * WAIT FOR NEXT SAMPLE TIMEOUT
        PAGE
*****************************************************************
*  This is the software automatic gain control factor update.  *
*  The routine keeps a running average plus three baud max's    *
*  to generate each new AGC update.  Once the value is gained  *
*  the routine uses a table lookup devide to force the filter  *
*  data max's into a tight range.                               *
*****************************************************************
AGCUPT ZALH AGCRA          ADD THE NEW BSMAX VALUE
       ADD  BSMAX,14       TO THE RUNNING AVERAGE
       SACH AGCRA          AND SAVE IT
       LAC  AGCNT          DECREMENT RUNNING AVERAGE COUNT
       SUB  ONE            SAVE IT AND
       SACL AGCNT          CHECK FOR ZERO
       SACH BSMAX          ZERO OUT RUNNING SIGNAL MAX
       BZ   OVROUT         IF ZERO THEN UPDATE AGC
       RET                 ELSE RETURN TO CALLING SEQUENCE
OVROUT LACK 3              RESET RUNNING AVERAGE COUNT
       SACL AGCNT          TO THREE
       LAC  AGCRA          MOVE AGCRA
       SACL AGCLEV         TO THE CALCULATION LEVEL
       LAC  AGCRA,14       DIVIDE RUNNING AVERAGE SUM
       SACH AGCRA          BY 4 TO GET NEW RUNNING AVERAGE
       LAC  AGCLEV         GET AVERAGE MAX SIGNAL LEVEL
       SUB  ONE,14         COMPARE TO 16384
       BLZ  ASHF1          IF LESS THAN SHIFT TABLE LOOKUP
       LAC  AGCLEV,7       GET LOOKUP VALUE
       SACH TEMP           MOVE LOOKUP VALUE TO
       LAC  TEMP           THE LOW HALF OF THE ACC
       ADD  AGCOFF         ADD IN TABLE OFFSET
       TBLR AGC            AND GET AGC VALUE
       LAC  AGC,15         DIVIDE THE AGC VALUE
       SACH AGC            BY 2 TO FORCE TO Q14 MODE
       RET                 RETURN TO CALLING SEQUENCE
ASHF1  ADD  ONE,13         COMPARE TO 8192
       BLZ  ASHF2          IF LESS THAN SHIFT TABLE LOOKUP
```

```
         LAC   AGCLEV,8     GET LOOKUP VALUE
         SACH  TEMP         MOVE LOOKUP VALUE TO
         LAC   TEMP         THE LOW HALF OF THE ACC
         ADD   AGCOFF       ADD IN TABLE OFFSET
         TBLR  AGC          AND GET AGC VALUE
         RET                RETURN TO CALLING SEQUENCE
ASHF2    ADD   ONE,12       COMPARE TO 4096
         BLZ   ASHF3        IF LESS THAN SHIFT TABLE LOOKUP
         LAC   AGCLEV,9     GET LOOKUP VALUE
         SACH  TEMP         MOVE LOOKUP VALUE TO
         LAC   TEMP         THE LOW HALF OF THE ACC
         ADD   AGCOFF       ADD IN TABLE OFFSET
         TBLR  AGC          AND GET AGC VALUE
         LAC   AGC,1        AGC VALUE * 2 TO ADJUST
         SACL  AGC          FOR LOWER SIGNAL STRENGTH
         RET                RETURN TO CALLING SEQUENCE
ASHF3    ADD   ONE,11       COMPARE TO 2048
         BLZ   ASHF4        IF LESS THAN SHIFT TABLE LOOKUP
         LAC   AGCLEV,10    GET LOOKUP VALUE
         SACH  TEMP         MOVE LOOKUP VALUE TO
         LAC   TEMP         THE LOW HALF OF THE ACC
         ADD   AGCOFF       ADD IN TABLE OFFSET
         TBLR  AGC          AND GET AGC VALUE
         LAC   AGC,2        AGC VALUE * 4 TO ADJUST
         SACL  AGC          FOR LOWER SIGNAL STRENGTH
         RET                RETURN TO CALLING SEQUENCE
ASHF4    ADD   ONE,10       COMPARE TO 1024
         BLZ   ASHF5        IF LESS THAN SHIFT TABLE LOOKUP
         LAC   AGCLEV,11    GET LOOKUP VALUE
         SACH  TEMP         MOVE LOOKUP VALUE TO
         LAC   TEMP         THE LOW HALF OF THE ACC
         ADD   AGCOFF       ADD IN TABLE OFFSET
         TBLR  AGC          AND GET AGC VALUE
         LAC   AGC,3        AGC VALUE * 8 TO ADJUST
         SACL  AGC          FOR LOWER SIGNAL STRENGTH
         RET                RETURN TO CALLING SEQUENCE
ASHF5    ADD   ONE,9        COMPARE TO 512
         BLZ   ASHF5        IF LESS THAN SHIFT TABLE LOOKUP
         LAC   AGCLEV,12    GET LOOKUP VALUE
         SACH  TEMP         MOVE LOOKUP VALUE TO
         LAC   TEMP         THE LOW HALF OF THE ACC
         ADD   AGCOFF       ADD IN TABLE OFFSET
         TBLR  AGC          AND GET AGC VALUE
         LAC   AGC,4        AGC VALUE * 16 TO ADJUST
         SACL  AGC          FOR LOWER SIGNAL STRENGTH
         RET                RETURN TO CALLING SEQUENCE
ASHF6    ADD   ONE,5        COMPARE TO 32
         BLZ   NOEDT        LOST MINIMUM ENERGY LEVEL
         LAC   AGCLEV,13    GET LOOKUP VALUE
         SACH  TEMP         MOVE LOOKUP VALUE TO
         LAC   TEMP         THE LOW HALF OF THE ACC
         ADD   AGCOFF       ADD IN TABLE OFFSET
         TBLR  AGC          AND GET AGC VALUE
         LAC   AGC,5        AGC VALUE * 32 TO ADJUST
         SACL  AGC          FOR LOWER SIGNAL STRENGTH
         RET                RETURN TO CALLING SEQUENCE
NOEDT    LACK  >DF          PASSBAND SIGNAL TOOL LOW
         AND   RECST        DISABLE SIGNAL ENERGY DETECT
         SACL  RECST        AND CARRIER DETECT SIGNAL
         RET                RETURN TO CALLING SEQUENCE
*
         PAGE
HANGUP   B     WAIT
DUMXMT   EQU   $
         RET
SMARK    EQU   $
         RET
*
```

```
*****************************************************
*
GETDBT EQU $
       IN     XMTD,PA6     * GET NEW DIBIT
       LACK   >30
       AND    XMTD         * CHECK COMMAND BITS
       BZ     COMD         * IF ZERO SQT MODEM, IDLE
*
       LACK   COEF         * RECYCLE IF FINISHED
       SACL   XPTR
       DMOV   XIBUF1       * SHIFT UP THE FILTER
       DMOV   XIBUF0       * TO MAKE ROOM FOR
       DMOV   XQBUF1       * FOR THE NEW DATA VALUE
       DMOV   XQBUF0       * JUST INPUT
*
       LACK   3
       AND    XMTD         * NEW DIBIT FROM 7000
       ADD    OFSET1       * LOOKUP NEWPHASE
       TBLR   XNEWPH
       LAC    XOLDPH       * GET OLDPHASE.
       ADD    XNEWPH       * ADD NEW PHASE.
       AND    MASK3        * MASK WITH >0006.
       SACL   XOLDPH       * STORE BACK 'NEW' OLDPHASE.
       ADD    INDXPH       * LOOKUP I & Q INPUTS.
       TBLR   XIBUF0
       ADD    ONE
       TBLR   XQBUF0
       RET
*
*
DUMMY  CALL DEMOD          ATTEMPT DEMODULATION
       RET                 RETURN TO TASK MASTER
*
*
DEMODB EQU  $              MIDDLE OF THE BAUD
       LACK >FE            RESET THE CURRENT BAUD CLOCK
       AND  RECST          CORRECTION FLAG IN THE STATUS
       SACL RECST          REGISTER AND SAVE IT
*******************************************************
*****   DEMODUATE THE PASSBAND SIGNAL.            *****
*******-----------------------------------------*******
*****   RCVR. CARRIER SINE(COSINE) WAVE GENERATOR *****
*******************************************************
DEMOD  EQU  $
        LAC    RALPHA,8     * DELTA IS THE INCREMENT.
        SACH   TEMP         * ISOLATE INTEGER PORTION.
        LAC    TEMP
        ADD    OFSETO       * ADD INDEX TO SINE TABLE.
        TBLR   SINA         * SINE VALUE,  (Q15).
        LACK   >20
        ADD    TEMP
        AND    MASK2
        ADD    OFSETO       * ADD INDEX TO COSINE TABLE.
        TBLR   COSA         * COSINE VALUE, (Q15).
*
******************************************************
*
CONT1   LT     ISUM         * DEMOD. I CHANNEL
        MPY    COSA         * A=(Yi * cosA)/2
        PAC
        LT     QSUM
        MPY    SINA
        APAC                * A=(Yi * cosA)/2 + (Yq * sin A)/2
        SACH   RECI,1       * RECI= (Yi * cosA) + (Yq * sinA)
*
        LT     ISUM         * DEMOD. Q CHANNEL
        MPY    SINA
        PAC                 * A = (Yi * sinA)/2
        LT     QSUM
        MPY    COSA
```

```
        SPAC                   * A =[(Yi  * s
        SACH  RECQ,1           * RECQ = (Yi
*
*---MUST DETERMINE ENERGY FOR BAUD CLOC
*
        LT    RECI
        MPY   RECI          * FIND I**2
        PAC
        LT    RECQ
        MPY   RECQ          * FIND Q**2
        APAC
        SACH  ENRGY         * ENERGY = (I*'
*
*---MUST DETERMINE SIGN OF I AND Q FOR
*
        LAC   RECI             * DETERMINE :
        BGZ   DM1
        LAC   MINUS1
        B     DM2
DM1     LAC   PLUS1
DM2     SACL  SIGNI            * SAVE SIGN (
*
        LAC   RECQ             * DETERMINE '
        BGZ   DM3
        LAC   MINUS1
        B     DM4
DM3     LAC   PLUS1
DM4     SACL  SIGNQ
        RET                   RETURN TO CA
****************************************
* INOUT GET DIBIT FROM 7000 AND XMIT N
* TO THE 7000
****************************************
OUT     EQU   $
        LAC   RDIBIT,10
        OR    MINUS1         * MASK D15-D1
        SACL  BITOUT         *AND SAVE THE
        OUT   BITOUT,PA6     *XMIT TO 7000
        RET                  * BACK TO CAL
********--------------------------------
*****     PHASE DECODING - BINARY TO GR
* THIS ROUTINE CALCULATES PHASE SHIFT
* CURRENT ABSOLUTE PHASE, GREY CODE RE
********--------------------------------
DECODE  LAC   RECI          * DETERMINE ABS
        BGZ   ABS1
        LAC   RECQ
        BGZ   ABS2
        LACK  2             * PHASE IS 2 (0
        B     DIFFER        *
ABS2    LACK  3             * PHASE IS 3 (2
        B     DIFFER        *
ABS1    LAC   RECQ
        BGZ   ABS3
        LACK  1             * PHASE IS 1 (1
        B     DIFFER        *
ABS3    LACK  0             * PHASE IS 0 (9
*                           *
DIFFER  SACL  TEMP
        SUB   ROLDPH        * SUBTRACT PRE\
        BGEZ  DF1           * LUTE PHASE (E
        ADD   FOUR
DF1     ADD   RPHSE         * MAP PHASE CH/
        TBLR  RDIBIT
        LAC   TEMP
        SACL  ROLDPH
********--------------------------------
*****     COMPUTE CARRIER ERROR SIGNAL.
*****     e(t) = RECI*SIGNQ - RECQ*SIGNI
********--------------------------------
```

```
COMERR ZAC
       LT      RECI
       MPY     SIGNQ
       LTA     RECQ
       MPY     SIGNI
       SPAC
       SACH    ERROR,1      * ERROR IS IN Q12
********-------------------------------------------********
*****   LOOP FILTER                                 ******
********-------------------------------------------********
       ZAC
       LT      PLL2
       MPY     ERROR
       LTA     PLL1
       MPY     ERRSIG
       APAC
       SACH    ERRSIG,1     * ERRSIG IS IN Q12
**********************************************************
* CORRECT PHASE ERROR ONLY AT MIDDLE OF BAUD             *
**********************************************************
*   Adjust carrier phase +/-                              *
*   one table entry if       - (2*trshld) > error > trshld   *
*   two table entries if     - (2*trshld) < error >> trshld  *
*   RALPHA is current local carrier table index.(in MSB )    *
**********************************************************
CKEROR  LAC   ERRSIG       *
        BGZ   ERR1         * If error is -ve add threshold
        ADD   TRSHD1       *
        BGZ   ERRETN       * Still -ve?... add again
        ADD   TRSHD1       *
        BGZ   SUB1A        * still -ve?...
        LAC   RALPHA       * Error >> trshld; add 2 to index
        SUB   ONE,9        *
        B     ERR2         *
SUB1A   LAC   RALPHA       * Error > trshld; add 1 to index
        SUB   ONE,8        *
        B     ERR2         *
ERR1    SUB   TRSHD1       * Error ia +ve; subtract threshold
        BLZ   ERRETN       * Error  > trshld
        SUB   TRSHD1       * see if error >> trshld
        BLZ   ADD1A        * No...add one to index
        LAC   RALPHA       * Yes...add 2 to index
        ADD   ONE,9        * SUB 2 same as ADD >7E in modulo 128
        B     ERR2         *
ADD1A   LAC   RALPHA       *
        ADD   ONE,8        *
ERR2    AND   MASK1        * Keep RALPHA modulo 128
        SACL  RALPHA       * save new index
        RET                * Return with corrected RALPHA
ERRETN  LAC   ONE,4        *
        OR    RECST        * If |error| less than threshold
        SACL  RECST        * set flag in status register
RETA    RET
**********************************************************
*****     BAUD  CLOCK  ALLIGNMENT                     ****
**********************************************************
BDCLK1 CALL  DEMOD
       LAC   ENRGY         * ENRGY = E(3)
       SACL  PENRGY        * STORE IT IN PENRGY
       RET
BDCLK2 CALL  DEMOD
       LAC   RECST         * TEST IF CORRECTION OF THE
       AND   ONE           * BAUD CLOCK IS MADE
       BNZ   RETB          * IF SO THEN RETURN
       LAC   ENRGY         * ENRGY = E(11), PENRGY = E(3)
       SUB   PENRGY        * FORM ERROR SIGNAL
       SACL  BERROR        * BERROR = E(11)-E(3)
********----------------------------------------**********
****     LOOP  FILTER                              *******
```

```
********--------------------------------------------------**********
        ZAC
        LT      BPLL2
        MPY     BERROR
        LTA     BPLL1
        MPY     BEROUT
        APAC
        SACH    BEROUT,1          * BEROUT IN Q14
*
*---APPLY  CORRECTION
*
        LAC     BEROUT
        BGEZ    POS               * TEST BERROUT SIGN.
        ADD     TRSHD2
        BGEZ    RETB              * IF |BERROUT|<TRSHD RETURN.
        ADD     TRSHD2            * BERROUT IS NEGATIVE. THEREFORE
        BGEZ    SUB1B             * ADJUST CLOCK BY DELAYING SAMPLE COUNT.
        LAC     SAMPLE            * IF |BERROUT|>2*TRSHD
        SUB     ONE,1             * MAKE TWO SAMPLE ADJUSTMENT
        SACL    SAMPLE            * OF THE SAMPLE (BAUD CLOCK)
        B       RETB              * COUNT.
SUB1B   LAC     SAMPLE            * IF TRSHD<|BERROUT|<2*TRSHD
        SUB     ONE               * MAKE ONE SAMPLE ADJUSTMENT
        SACL    SAMPLE            * OF THE SAMPLE (BAUD CLOCK)
        B       RETB              * COUNT.
POS     SUB     TRSHD2            * BERROUR IS POSITIVE. THEREFORE
        BLZ     RETB              * ADJUST CLOCK BY ADVANCING SAMPLE
        SUB     TRSHD2            * COUNT.
        BLZ     ADD1B             * IF |BERROUT|>2*TRSHD
        LAC     SAMPLE            * MAKE TWO SAMPLE ADJUSTMENT
        ADD     ONE,1             * OF THE SAMPLE (BAUD CLOCK)
        SACL    SAMPLE            * COUNT.
        B       RETB
ADD1B   LAC     SAMPLE            * IF TRSHD<|BERROUT|<2*TRSHD
        ADD     ONE               * MAKE ONE SAMPLE ADJUSTMENT
        SACL    SAMPLE            * OF THE SAMPLE (BAUD CLOCK) COUNT.
RETB    LAC     RECST             * SET FLAG TO INDICATE THAT THE BAUD
        OR      ONE               * CLOCK ADJUSTMENT IS MADE.
        SACL    RECST
        RET
*******************************************************************************
        END
```

Appendix E
TMS7742 Source Code

```
         TITL       TMS 7742 MODEM INTERFACE PROGRAM'
         OPTION   XREF,TUNLST
 *                7042 PORT ASSIGNMENTS
 *
 * APORT
 *             A7    A6    A5    A4    A3    A2    A1    A0
 *             OHR_  N.C.  RCVD  ATE_  A_/O  SQT   DTR   DCD
 *             (O)   (X)   (I)   (O)   (O)   (O)   (I)   (I)
 *
 * BPORT
 *             B7    B6    B5    B4    B3    B2    B1    B0
 *             NB8   NB4   NB2   NB1   TXD   DP    DSR   CTS
 *             (O)   (O)   (O)   (O)   (O)   (O)   (O)   (O)
 *
 * CPORT
 *             C7    C6    C5    C4    C3    C2    C1    C0
 *             ACKW  ACKR  CMD2  CMD1  TDB3  TDB2  TDB1  TDB0
 *             (O)   (O)   (O)   (O)   (O)   (O)   (O)   (O)
 *
 * DPORT
 *             D7    D6    D5    D4    D3    D2    D1    D0
 *             NEWO  NEWI  CDT   ENB   RDB3  RDB2  RDB1  RDB0
 *             (I)   (I)   (I)   (I)   (I)   (I)   (I)   (I)
 *
 *----------------------------------------------------------------
 *             +-------------------------------+
 * SWSTAT      | 7 | 6 | 5 | 4 | 3 | 2 | 1 | 0 |
 *             +-------------------------------+
 *
 * BIT7: modem type        0= B103 mode
 *                         1= B212 mode
 *
 * BIT6: timer flag        0= carrier wait timer enabled
 *                         1= 1200 Hz timer enabled
 *
 * BIT5: 1st dibit flag    0= flag reset
 *                         1= flag set
 *----------------------------------------------------------------
           IDT 'DSPMODM'
 *         OPTION   XREF
           AORG     >F006
 *
 * 7041 Peripheral Memory Symbols
 *
IOCNT0 EQU P0
T1DATA EQU P2
T1CNTL EQU P3
APORT  EQU P4
ADDR   EQU P5
BPORT  EQU P6
CPORT  EQU P8
CDDR   EQU P9
DPORT  EQU P10
DDDR   EQU P11
IOCNT1 EQU P16
SMODE  EQU P17
SCTL0  EQU P17
SSTAT  EQU P17
T2DATA EQU P18
T2CNTL EQU P19
T3DATA EQU P20
SCTL1  EQU P21
RXBUF  EQU P22
TXBUF  EQU P23
MPRTC  EQU >108
MPRTD  EQU >10A
 *
 * Bit Masks.
 *
```

```
BIT0    EQU >01
BIT1    EQU >02
BIT2    EQU >04
BIT3    EQU >08
BIT4    EQU >10
BIT5    EQU >20
BIT6    EQU >40
BIT7    EQU >80
*
NOT0    EQU >FE
NOT1    EQU >FD
NOT2    EQU >FB
NOT3    EQU >F7
NOT4    EQU >EF
NOT5    EQU >DF
NOT6    EQU >BF
NOT7    EQU >7F
*
* Ascii constants
*
TAB     EQU >09                         ; tab character
BLANK   EQU >20                         ; space character
COMMA   EQU >2C                         ; ','
LF      EQU 10
CR      EQU 13
BS      EQU 8                           ; BACKSPACE CHARACTER
POUND   EQU >23                         ; '#'
STAR    EQU >2A                         ; '*'
*
ISA     EQU >41                         ; 'A'
ISZ     EQU >5A                         ; 'Z'
*
* 7041 RAM map
*
RDIBIT EQU R2
DIBIT2 EQU RDIBIT+1                     ; receiver input sequence
DIBIT1 EQU DIBIT2+1                     ; from the 32010
TEMP1  EQU DIBIT1+1                     ; temporary register (receiver)
TEMP2  EQU TEMP1+1                      ; temporary register (xmitter)
RBIT14 EQU TEMP2+1
RBIT17 EQU RBIT14+1
DESREG EQU RBIT17+1
FLAG   EQU DESREG+1
COUNTE EQU FLAG+1
BITCNT EQU COUNTE+1
CHRCNT EQU BITCNT+1
TRNMIT EQU CHRCNT+1                     ; character to be transmitted
STPFLG EQU TRNMIT+1                     ; stop bit deleted flag
MCOUNT EQU STPFLG+1                     ; mark counter
XDBIT2 EQU MCOUNT+1
XDBIT1 EQU XDBIT2+1
XBIT17 EQU XDBIT1+1
XBIT14 EQU XBIT17+1
SCMREG EQU XBIT14+1
XDIBIT EQU SCMREG+1                     ; xmitter input dibit
CONTER EQU XDIBIT+1                     ; counter register
RBTCNT EQU CONTER+1                     ; character bit counter (xmt)
XMTCHR EQU RBTCNT+1                     ; input character buffer
ADDRES EQU XMTCHR+2                     ; command buffer address pointer
PNTR   EQU ADDRES+1                     ; counter register
SWSTAT EQU PNTR+1                       ; software status flag
LOCHI  EQU SWSTAT+1
LOCLO  EQU LOCHI+1
ADDR1  EQU LOCLO
INDEX1 EQU ADDR1+1                      ; general purpose register
INDEX2 EQU INDEX1+1                     ; general purpose register
COUNT1 EQU INDEX2+1
COUNT  EQU COUNT1+1                     ; general use double register counter
```

```
COMBUF EQU COUNT+1                         ; beginning of the command buffer
S0     EQU COMBUF+40                       ; carriage return register
S1     EQU S0+1                            ; line feed register
S2     EQU S1+1                            ; backspace register
S3     EQU S2+1                            ; # of rings to answer on
S4     EQU S3+1                            ; # of rings detected
S5     EQU S4+1                            ; escape code character
INT5TM EQU S5+1                            ; interrupt 5 timer register
VALUE  EQU INT5TM+1                        ; contains numerical value of parameters
MSGM   EQU VALUE+1
MSGL   EQU MSGM+1                          ; masseage address register
CWT1   EQU MSGL+1
CWT2   EQU CWT1+1                          ; carrier wait abort timer
MSTIME EQU CWT2+2                          ; millisec timing register
DELYR1 EQU MSTIME+1
STACK  EQU R100
*
ALL    EQU >FF
ZERO   EQU >00
ONE    EQU >01
TWO    EQU >02
THREE  EQU >03
EIGHT  EQU >08
NINE   EQU >09
TEN    EQU >0A
CNTVAL EQU >DC
ADDTOP EQU S0-1
ADDBOT EQU COMBUF
*******-------------------------------------*******
*****             Initialization                *****
*******-------------------------------------*******
INIT   MOVP   %>0F,IOCNT0          ; s.c. mode, enable int2 and int1
       MOVP   %>00,IOCNT1          ; disable int4 and int5
       MOVP   %>0C,APORT
       MOVP   %>9C,ADDR            ; set direction of APORT
       MOVP   %>FB,BPORT
       MOVP   %>CF,CPORT
       MOVP   %>FF,CDDR            ; set direction of CPORT
       MOVP   %>CF,DPORT
       MOVP   %>00,DDDR            ; set direction of DPORT
       MOVP   %155,T2DATA
       MOVP   %>81,T2CNTL
       ORP    %BIT2,BPORT          ; reset the 99531 dialer
       ANDP   %NOT2,BPORT
       MOV    %STACK,B
       LDSP                        ; load stack pointer
       MOV    %ALL,DIBIT1
       MOV    %ALL,DIBIT2
       MOV    %ONE,RBIT17
       MOV    %ONE,RBIT14
       MOV    %ALL,XDBIT1
       MOV    %ALL,XDBIT2
       MOV    %ONE,XBIT17
       MOV    %ONE,XBIT14
*
       MOV    %CR,S0               ; carriage return character
       MOV    %LF,S1               ; line feed character
       MOV    %BS,S2               ; backspace character
       MOV    %ONE,S3              ; # of rings to answer on
       CLR    S4                   ; # of rings detected
       MOV    %'+',S5              ; escape code character
       MOV    %>C0,SWSTAT          ; software flag default conditions
*
* main routine
*
       ANDP   %NOT0,BPORT          ; set CTS_
       EINT
*
```

```
       CALL    @AUTOBD              ; Autobaud to terminal speed
TOP    MOVD    %HELLO,MSGL          ; Send hello message
       CALL    @PRINT
*
* look for input commands.
*
LOOK   CALL    @CLEAR               ; clear the command buffer
       MOVD    %ADDTOP,ADDRES       ; point to top of the buffer
       CLR     PNTR                 ; clear buffer command pointer
LK4COM BTJZP   %BIT1,SSTAT,LK4COM ; command received?
       MOVP    RXBUF,A
       MOVP    A,TXBUF              ; echo
WAIT4  BTJZP   %BIT2,SSTAT,WAIT4
       CMP     %CR,A                ; last character?
       JEQ     EXEC                 ; yes, go execute command
*
       CMP     %'(',A               ; ignore
       JEQ     LK4COM
       CMP     %')',A               ; ignore
       JEQ     LK4COM
       CMP     %'-',A               ; ignore
       JEQ     LK4COM
       CMP     %' ',A               ; ignore
       JEQ     LK4COM
       CMP     %'/',A               ; ignore
       JEQ     LK4COM
       CMP     %BS,A                ; backspace?
       JNE     NXTSTG               ; yes, go get new command
       DEC     PNTR                 ; decrement pointer
       CLR     A                    ; CLEAR OUT THE BUFFER
       STA     *ADDRES              ; AT THE CURRENT LOCATION
       INC     ADDRES               ; point to the previous location
       JMP     LK4COM
*
NXTSTG INC     PNTR                 ; command buffer pointer
       STA     *ADDRES              ; location for command
       DECD    ADDRES               ; location for next command
       CMP     %40,PNTR             ; allow 40 chars maximum
       JEQ     ERR                  ; more than 40..clear buffer
       JMP     LK4COM               ; keep going till <CR>
*
ERR    CALL    @CLEAR               ; clear command buffer
       MOVD    %ERROR,MSGL          ; send error message
       CALL    @PRINT
       MOV     %STACK,B             ; reset the stack pointer
       LDSP
       BR      @LOOK
*
EXEC     MOVD  %ADDTOP,ADDRES       ; initialize address point
         LDA   *ADDRES              ; get command
         CMP   %'A',A
         JL    ERR                  ; Check for A thru Z
         CMP   %'Z'+1,A
         JHS   ERR
         CLR   B                    ; Parameter buffer pointer
         DECD  ADDRES
         SUB   %'A',A
         MOV   A,B
         RL    B                    ; B*2
         LDA   @COMLIS(B)
         MOV   A,LOCHI              ; MSB address
         INC   B
         LDA   @COMLIS(B)
         MOV   A,LOCLO              ; LSB address
         BR    *ADDR1               ; execute command
*
         PAGE
```

```
*******----------------------------------------*******
*****      Local Digital Loopback Test            *****
*******----------------------------------------*******
*
LDLB    EQU     $
        MOVD    %LDLBM,MSGL          ; RESPOND TO COMMAND TO DTE
        CALL    @PRINT               ; BY PRINTING TEST CODE
        MOV     %>10,R23             ; SET COMMAND TO LDLB MODE
        BR      @GO320               ; AND RUN THE 320
*
        PAGE
*******----------------------------------------*******
*****               Dial Blind                    *****
*******----------------------------------------*******
*
DB      OR      %BIT6,SWSTAT
        MOVP    %>2A,IOCNTO           ; disable RI interrupt
        ORP     %BITO,BPORT           ; turn off CTS_
        ORP     %>8C,APORT            ; originate mode, squelch 532,
*                                          and go off hook
        CALL    @DIAL                 ; dial
        MOV     %18,CWT1              ; initialize carrier abort timer.
        CLR     CWT2
        ORP     %BIT2,IOCNTO          ; enable carrier abort interrupt
CHKDCD  BTJOP   %BITO,APORT,CHKDCD    ; wait for DCD_
        AND     %NOT6,SWSTAT
        BTJZ    %BIT7,SWSTAT,B103     ; check for modem type
        ORP     %BIT3,CPORT           ; 32010 in B212 originate mode
        ORP     %BIT2,CPORT
        BR      @B212
*******----------------------------------------*******
*****        Bell 103 Call Initiation.            *****
*******----------------------------------------*******
B103    BTJOP   %BITO,APORT,B103      ; Wait for DCD_
*
* Send originate tone.
*
        ORP     %BIT4,APORT           ; ATE = 1
        ANDP    %NOT2,APORT           ; unsquelch 532.
*
        MOVP    %>4A,IOCNTO           ; got DCD_, disable abort interrupt
*
* Wait 800ms
*
        MOVD    %800,MSTIME
        CALL    @MSDLY
        ANDP    %NOTO,BPORT           ; activate CTS_
        MOVD    %CONN3,MSGL           ; send connect 300 message
        CALL    @PRINT
        BR      @DAT103               ; enter data mode
*******----------------------------------------*******
*****   DIAL - Dial number stored in ADDRES.      *****
*******----------------------------------------*******
DIAL    ANDP    %NOT4,APORT           ; ATE_ = 0, enable EXI mode
*
* Execute dialing.
*
        MOVD    %4000,MSTIME          ; Initial dial tone wait of 2 second
        CALL    @MSDLY
NXTDIG  LDA     *ADDRES               ; Load subcommand
        CMP     %ZERO,A               ; is it the last command?
        JNE     NOTEND
*
* End of dialing.
*
        RETS
*
```

```
* Case statement to determine subcommand.
*
NOTEND DECD   ADDRES               ; update address
       CMP    %'0',A               ; check less than '0'
       JL     NOTNUM
       CMP    %'9'+1,A             ; check greater than '9'
       JHS    NOTSPC
       BR     @ISANUM
NOTNUM CMP    %',',A               ; ',' - dial tone wait
       JEQ    DPAUSE
       CMP    %STAR,A              ; '*' - tone dial *
       JEQ    ISSTAR
       CMP    %POUND,A             ; '#' - tone dial #
       JEQ    APOUND
NOTSPC BR     @NXTDIG
*
* Wait for a dial tone.
*
DPAUSE MOV    %TWO,VALUE           ; Blind delay
       CALL   @SECDLY
       BR     @NXTDIG
*
* Dial a digit.
*
ISSTAR MOV    %TEN,A               ; dial * if tone dial
       JMP    OUTDIG
APOUND MOV    %11,A                ; dial # if tone dial
       JMP    OUTDIG
ISANUM SUB    %'0',A               ; dial a number
OUTDIG ANDP   %>0F,BPORT           ; clear old digit
       RL     A                    ; get the correct value
       RL     A
       RL     A
       RL     A
       ORP    A,BPORT              ; send new digit
PNDWT0 BTJZP  %BIT1,APORT,PNDWT0 ; wait for acceptance
       ORP    %BIT2,BPORT          ; set DP
PNDWT1 BTJOP  %BIT1,APORT,PNDWT1 ; wait for PND low
       ANDP   %NOT2,BPORT          ; clear DP
PNDWT2 BTJZP  %BIT1,APORT,PNDWT2 ; wait for PND high
       BR     @NXTDIG
*
       PAGE
*******---------------------------------------*******
*****      BELL 1200 BPS MODEM ALGORITHM         *****
*******---------------------------------------*******
*
B212   EQU    $
       MOV    %>20,R23           ; SET COMMAND TO MODEM RUN
GO320  CLR    R2                 ; CLEAR COM STATUS REG
       CLR    R11                ; INITIALIZE SCRAMBLER HISTORY
       CLR    R12                ; AS ALL ZEROS
       CLR    R18                ; INITIALIZE DESCRAMBLER HISTORY
       CLR    R19                ; AS ALL ZEROS
       CLR    R13                ; INITIALIZE DESCRAMBLER HISTORY
       CLR    R20                ; AS ALL ZEROS
       ANDP   %NOT0,BPORT        ; ACTIVATE CTS TO DTE
       CLR    A                  ; CYCLE THE CLEAR LINES
       STA    @MPRTC             ; OF THE I/O CONTROL
       ORP    %>C0,CPORT         ; RESET 320 ACK LINES
*
*    START UP MODEM OR DLB TEST
*
STOPB2 MOV    %3,R10             ; SET DIBIT TO MARKS
       CALL   @SCRAM             ; AND SCRAMBLE IT
       MOV    R10,A              ; HOLD IT FOR TRANSMIT
       OR     R23,A              ; OR IN COMMAND BITS
```

```
       ANDP   %>C0,CPORT          ; CLEAR OFF CURRENT BITS
       ORP    A,CPORT             ; SEND OUT SCRAM MARKS
*  TRANSMIT UNSCRAMBLED MARKS AND RECEIVE
MRC1   BTJZP  %BIT7,DPORT,MRC2    ; WAIT FOR WRITE FROM 320
CHKTCH BTJOP  %BIT6,DPORT,RECDTE  ; WAIT FOR READ FROM 320
       BR     @MRC3               ; PROCESS READ FROM 320
RECDTE BTJOP  %BIT1,SSTAT,DTEGET  ; IS DTE REC BUF FULL
XMTDTE BTJOP  %BIT0,SSTAT,DTEPUT  ; IS DTE TRANS BUF EMPTY
       JMP    MRC1                ; LOOK AGAIN
*
*     CODE INTERFACE TO DTE
*
DTEGET EQU    $
       MOVP   RXBUF,A             ; YES, GET THE CHARACTER?
       CMP    %>1B,A              ; IF A <> ESCAPE
       JMP    OVRSQT              ; THEN CONTINUE
       CLR    A                   ; ELSE SQUELCH THE
       STA    @MPRTC              ; THE 320 MODEM AND
       BR     @TOP                ; AND RETURN TO MONITOR
OVRSQT INC    R24                 ; INCREMENT BYTE COUNT
       BTJO   %BIT5,R2,DTEGER     ; CHECK FOR BUF2 FULL
       BTJO   %BIT3,R2,DTEG1      ; CHECK IF 1ST CHAR
       OR     %BIT7,R2            ; FLAG FOR START BIT
       MOV    A,R7                ; IF SO THEN RESTART
       MOV    %>A,R21             ; RESET XMT COUNT
       OR     %BIT3,R2            ; SET TRANS ACTIVE
       JMP    XMTDTE              ; CHECK OUTPUT
DTEG1  MOV    A,R28               ; SAVE IT IN THE BUF2
       OR     %BIT5,R2            ; SET BUF2 FULL FLAG
       JMP    XMTDTE              ; CHECK OUTPUT
*
DTEGER CLR    A                   ; SQUELCH THE
       STA    @MPRTC              ; 320 MODEM
       MOVD   %BUFERR,MSGL        ; SEND ERROR MESSAGE
       CALL   @PRINT              ; TO USER TERMINAL
       BR     @TOP                ; EXIT ROUTINE
*
DTEPUT EQU    $
       BTJZ   %BIT4,R2,MRC1       ; CHECK FOR CHARACTER READY
       MOV    R29,A               ; GET BUFFERED CHARACTER
       MOVP   A,TXBUF             ; SEND IT TO THE DTE
       AND    %NOT4,R2            ; RESET BUFFER FULL FLAG
       JMP    MRC1                ; RETURN TO FLAG LOOP
*
       PAGE
*
*     RECEIVE DIBITS FROM THE 320
*
MRC2   ANDP   %>7F,CPORT          ; RESET WRITE ACKNOWLEDGE
       ORP    %>80,CPORT          ; BY TOGGLING LINES
       MOVP   DPORT,A             ; GET THE RETURNED DATA
       MOV    A,R10               ; AND HOLD IT IN R10
       BTJZ   %BIT5,A,CHKTCH      ; IF NO CARRIER THEN DONE
       AND    %3,R10              ; AND OFF STATUS
       CALL   @DSCRAM             ; DESCRAMBLE IT
       BTJO   %BIT2,R2,RCHAR1     ; CHECK FOR REC CHAR ACTIVE
       RRC    R10                 ; CHECK DIBIT0
       JC     RNB                 ; IF HIGH THEN CHECK NEXT
       RRC    R10                 ; SAVE LSB OF RECEIVE CHAR
       RRC    R5                  ; IN CHAR HOLD REG
       MOV    %7,R22              ; SET REC BIT COUNT REG
       JMP    RCHAR0              ; SKIP OVER NEXT CHECK
*
RNB    RRC    R10                 ; CHECK DIBIT1
       JC     CHKTCH              ; IF HIGH THEN CHECK XMTCHAR
       MOV    %8,R22              ; SET REC BIT COUNT REG
RCHAR0 OR     %BIT2,R2            ; SET REC CHAR ACTIVE
       BR     @RECDTE             ; CHECK DTE
```

```
*
RCHAR1 SUB    %2,R22              ; CHECK BIT POSITION
       JP     RCHAR3              ; IF > 0 GET 2 BITS
       JZ     RCHAR2              ; IF = 0 GET 1 BIT
       RRC    R10                 ; PUT BIT7 INTO
       RRC    R5                  ; REC CHAR HOLD REG
       MOV    R5,R29              ; PUT CHAR IN OUT BUFFER
       OR     %BIT4,R2            ; SET BUFFER FULL FLAG
       CLR    R5                  ; CLEAR BUFFER FOR NEXT CHAR
       AND    %NOT2,R2            ; RESET REC CHAR ACTIVE
       JMP    RNB                 ; CHECK DIBIT1 FOR START BIT
*
RCHAR2 RRC    R10                 ; SAVE MSB OF RECEIVE CHAR
       RRC    R5                  ; INTO REC CHAR HOLD REG
       RRC    R10                 ; PUT BIT7 INTO
       RRC    R5                  ; REC CHAR HOLD REG
       MOV    R5,R29              ; PUT CHAR IN OUT BUFFER
       OR     %BIT4,R2            ; SET BUFFER FULL FLAG
       CLR    R5                  ; CLEAR BUFFER FOR NEXT CHAR
       AND    %NOT2,R2            ; RESET REC CHAR ACTIVE
       BR     @RECDTE             ; CHECK DTE
*
RCHAR3 RRC    R10                 ; MOVE DIBIT0 TO
       RRC    R5                  ; REC CHAR HOLD REG
       RRC    R10                 ; MOVE DIBIT1 TO
       RRC    R5                  ; REC CHAR HOLD REG
       BR     @RECDTE             ; CHECK DTE
*
       PAGE
*
*   SEND DIBITS TO THE 320
*
MRC3   ANDP   %>BF,CPORT          ; RESET ACKNOWLEDGE
       ORP    %>40,CPORT          ; BY TOGGLING LINES
       BTJO   %BIT3,R2,TCHAR0     ; CHECK FOR TRANS CHAR ACTIVE
       BR     @STOPB2             ; IF NOT SEND STOPBITS
TCHAR0 CLR    R10                 ; CLEAR OUT DIBIT REG
       SUB    %2,R21              ; CHECK POSITION
       JP     TCHAR6              ; > 2 MEANS TRANSMIT BITS
       JNZ    TCHAR3              ; IF PATTERN ONE THEN ODD
       RRC    R7                  ; GET BIT 7 FROM CHAR
       JNC    TCH00               ; IF NO CARRY DIBIT0=0
       OR     %BIT0,R10           ; ELSE DIBIT0=1
TCH00  BTJO   %BIT5,R2,TCHAR1     ; IF BUF2 EMPTY
       AND    %NOT3,R2            ; RESET TRAN ACTIVE BIT
       OR     %BIT1,R10           ; SET DIBIT1 TO STOP
       JMP    TCHSND              ; AND SEND DIBIT
TCHAR1 CMP    %9,R24              ; CHECK CHAR COUNT
       JL     TCHAR2              ; IF < DON'T DELETE STOPBIT
       CLR    R24                 ; CLEAR BYTE COUNT
       AND    %NOT1,R10           ; SEND DIBIT1 TO START
       MOV    R28,R7              ; LOAD IN NEW CHAR
       MOV    %9,R21              ; SET BIT COUNT
       AND    %NOT5,R2            ; RESET BUF2 FULL FLAG
       JMP    TCHSND              ; SEND THE DIBIT
TCHAR2 OR     %BIT1,R10           ; SEND DIBIT1 TO STOP
       MOV    R28,R7              ; LOAD IN NEW CHAR
       OR     %BIT7,R2            ; FLAG IN START BIT
       MOV    %>A,R21             ; SET BIT COUNT
       AND    %NOT5,R2            ; RESET BUF2 FULL FLAG
       JMP    TCHSND              ; SEND THE DIBIT
TCHAR3 BTJO   %BIT5,R2,TCHAR4     ; IF BUF2 EMPTY
       AND    %NOT3,R2            ; RESET TRAN ACTIVE BIT
       BR     @STOPB2             ; AND SEND MARKS
TCHAR4 CMP    %9,R24              ; CHECK CHAR COUNT
       JL     TCHAR5              ; IF < DON'T DELETE STOPBIT
       CLR    R24                 ; CLEAR BYTE COUNT
```

```
        MOV     R28,R7                  ; LOAD IN NEW CHAR
        OR      %BIT7,R2                ; FLAG IN START BIT
        MOV     %8,R21                  ; SET BIT COUNT
        AND     %NOT5,R2                ; RESET BUF2 FULL FLAG
        JMP     TCHAR6                  ; SEND THE DIBIT
TCHAR5  MOV     %1,R10                  ; SEND STOP THEN START
        MOV     R28,R7                  ; LOAD IN NEW CHAR
        MOV     %9,R21                  ; SET BIT COUNT
        AND     %NOT5,R2                ; RESET BUF2 FULL FLAG
        JMP     TCHSND                  ; SEND THE DIBIT
TCHAR6  BTJZ    %BIT7,R2,TCHAR7         ; START BIT NEEDED
        AND     %NOT7,R2                ; RESET START BIT FLAG
        JMP     TCHO1                   ; SKIP DIBIT1
TCHAR7  RRC     R7                      ; GET NEXT BIT OF CHAR
        JNC     TCHO1                   ; IF LOW SKIP BIT SET
        OR      %1,R10                  ; ELSE SET DIBIT0 TO ONE
TCHO1   RRC     R7                      ; GET NEXT BIT OF CHAR
        JNC     TCHSND                  ; IF LOW SKIP BIT SET
        OR      %2,R10                  ; ELSE SET DIBIT1 TO ONE
TCHSND  EQU     $
        CALL    @SCRAM                  ; AND SCRAMBLE IT
        MOV     R10,A                   ; HOLD IT FOR TRANSMIT
        ANDP    %>F0,CPORT              ; CLEAR OUT DIBIT VALUE
        ORP     A,CPORT                 ; SEND TO PORT
        BR      @RECDTE                 ; WAIT FOR RETURN LOOP
*
        PAGE
*******---------------------------------------*******
*****         Receiver descrambler                *****
*****     X(N) = Y(N-17) XOR Y(N-14) XOR Y(N)     *****
*******---------------------------------------*******
DSCRAM EQU      $
*
        MOV     R10,B           ; SAVE SCRAMBLED DIBIT
        CLR     R16             ; CLEAR THE Y(N-14) REFERENCE
        CLR     R17             ; CLEAR THE Y(N-17) REFERENCE
        MOV     R11,A           ; GET THE DESCRAMBLER HISTORY
        RL      A               ; SHIFT OUT Y(N-18)
        RLC     A               ; GET HISTORY Y(N-17)
        RLC     R17             ; AND PUT INTO REFERENCE
        RLC     A               ; SHIFT OFF TWO MORE BITS
        RLC     R17             ; SAVE Y(N-16) REFERENCE
        RLC     A               ; TO GET TO THE Y(N-14)
        RLC     A               ; AND GET HISTORY
        RLC     R16             ; AND PUT INTO REFERENCE
        RLC     A               ; GET HISTORY Y(N-13)
        RLC     R16             ; AND PUT INTO REFERENCE
        XOR     R16,R10         ; R10=X(N) XOR Y(N-14)
        XOR     R17,R10         ; R10=X(N) XOR Y(N-14) XOR Y(N-17)
        CLRC                    ; CLEAR OUT THE CARRY BIT
        RRC     R10             ; REVERSE THE DIBITS FOR
        JNC     OVRSW1          ; ALLIGNMENT WITH SCRAMBLER
        OR      %2,R10          ; IF CARRY THEN BIT HIGH
OVRSW1  EQU     $
*
        RLC     R13             ; SHIFT UP THE LSB HISTORY BITS
        RLC     R12             ; AND CARRY TO CSB HISTORY BITS
        RLC     R11             ; AND CARRY TO MSB HISTORY BITS
        CLRC                    ; CLEAR THE CARRY BIT
        RLC     R13             ; SHIFT UP THE LSB HISTORY BITS
        RLC     R12             ; AND CARRY TO CSB HISTORY BITS
        RLC     R11             ; AND CARRY TO MSB HISTORY BITS
        RRC     B               ; GET DIBIT0 AND
        RRC     R13             ; AND SHIFT IT INTO R13
        RRC     B               ; GET DIBIT0 AND
        RRC     R13             ; AND SHIFT IT INTO R13
*
        RETS
```

```
*
       PAGE
*******-----------------------------------------*******
*****        Transmitter Scrambler               *****
*****     Y(N) = Y(N-17) XOR Y(N-14) XOR X(N)    *****
*******-----------------------------------------*******
SCRAM  EQU    $
       CLRC                  ; CLEAR OUT THE CARRY BIT
       RRC    R10            ; REVERSE THE DIBITS FOR
       JNC    OVRSW2         ; ALLIGNMENT WITH SCRAMBLER
       OR     %2,R10         ; IF CARRY THEN BIT HIGH
OVRSW2 EQU    $
       CLR    R16            ; CLEAR THE Y(N-14) REFERENCE
       CLR    R17            ; CLEAR THE Y(N-17) REFERENCE
       MOV    R18,A          ; GET THE SCRAMBLER HISTORY
       RL     A              ; SHIFT OUT Y(N-18)
       RLC    A              ; GET HISTORY Y(N-17)
       RLC    R17            ; AND PUT INTO REFERENCE
       RLC    A              ; SHIFT OFF TWO MORE BITS
       RLC    R17            ; SAVE Y(N-16) REFERENCE
       RLC    A              ; TO GET TO THE Y(N-14)
       RLC    A              ; AND GET HISTORY
       RLC    R16            ; AND PUT INTO REFERENCE
       RLC    A              ; GET HISTORY Y(N-13)
       RLC    R16            ; AND PUT INTO REFERENCE
       XOR    R16,R10        ; R10=X(N) XOR Y(N-14)
       XOR    R17,R10        ; R10=X(N) XOR Y(N-14) XOR Y(N-17)
*
       MOV    R10,B          ; HOLD SCRAMBLED DIBIT FOR HISTORY
       RLC    R20            ; SHIFT UP THE LSB HISTORY BITS
       RLC    R19            ; AND CARRY TO CSB HISTORY BITS
       RLC    R18            ; AND CARRY TO MSB HISTORY BITS
       CLRC                  ; CLEAR CARRY BIT
       RLC    R20            ; SHIFT UP THE LSB HISTORY BITS
       RLC    R19            ; AND CARRY TO CSB HISTORY BITS
       RLC    R18            ; AND CARRY TO MSB HISTORY BITS
       RRC    B              ; GET DIBITO AND
       RRC    R20            ; AND SHIFT IT INTO R20
       RRC    B              ; GET DIBITO AND
       RRC    R20            ; AND SHIFT IT INTO R20
*
       RETS
*
       PAGE
*******-----------------------------------------*******
***** MSDLY - Wait MSTIME number of milliseconds *****
*******-----------------------------------------*******
MSDLY  EQU    $
       MOV    %CNTVAL,DELYR1     ; load the inner counter (9)
HERE2  DJNZ   DELYR1,HERE2       ; (9+2)
       DECD   MSTIME             ; (11)
       JC     MSDLY              ; (7)
       RETS
*******-----------------------------------------*******
*****    SECDLY - Wait VALUE number of seconds    *****
*******-----------------------------------------*******
SECDLY CMP    %0,VALUE
       JEQ    NODLY
NXTSEC MOVD   %1001,MSTIME
       CALL   @MSDLY
       DJNZ   VALUE,NXTSEC
NODLY  RETS
*******-----------------------------------------*******
*****               PRINT  subroutine             *****
*******-----------------------------------------*******
* MSGM and MSGL contain the address of text to print
* for messages to the screen
*
```

```
PRINT   CALL    @CRLF
PRINT1  LDA     *MSGL
        JZ      WAIT6
        MOVP    A,TXBUF              ; print each character in text statement
WAIT5   BTJZP   %BIT0,SSTAT,WAIT5 ; wait for txbuf ready
        INC     MSGL
        ADC     %0,MSGM
        JMP     PRINT1
WAIT6   CALL    @CRLF
        RETS
*
* send carriage return/line feed
*
CRLF    MOV     S0,A
        MOVP    A,TXBUF              ; send carriage return
CRWAIT  BTJZP   %BIT0,SSTAT,CRWAIT
        MOV     S1,A
        MOVP    A,TXBUF              ; send line feed
LFWAIT  BTJZP   %BIT2,SSTAT,LFWAIT
        RETS
*
        PAGE
*******-----------------------------------------*******
*****                PRINT  subroutine             *****
*******-----------------------------------------*******
AUTOBD  EQU     $
        MOV     %>20,A               ; SET BAUD CLOCK FOR
        MOVP    A,T3DATA             ; FOR OVERSPEED DTE
SETMOD  MOVP    %0,P17               ; Write to P17 to guarantee
        MOVP    %>60,SCTL0           ; we are talking to SCTL0, then reset
*                                    ; serial port
        MOVP    B,SMODE              ;
        MOVP    %>15,SCTL0           ;
        MOVP    %>40,SCTL1           ;
*
        MOVP    %BIT6,SCTL0          ; Parity error, parity is disabled in DTE.
        MOVP    %>6E,SMODE           ; Disable parity of port
        MOVP    %>15,SCTL0
        MOVP    %>40,SCTL1
        RETS
*
*******-----------------------------------------*******
*****      screen messages - text statements       *****
*******-----------------------------------------*******
ERROR   TEXT    'ERROR'
        BYTE    0
*
BUFERR  TEXT    'DTE BUFFER OVERFLOW ERROR'
        BYTE    0
*
CONN12  TEXT    'CONNECT 1200'
        BYTE    0
*
CONN3   TEXT    'CONNECT 300'
        BYTE    0
*
NOCAR   TEXT    'NO CARRIER'
        BYTE    0
*
RCALL   TEXT    'RING'
        BYTE    0
*
RESET   TEXT    'OK'
        BYTE    0
*
LDLBM   TEXT    'EXECUTE LDLB, ENTER CHARACTERS'
        BYTE    0
*
```

```
IALBM  TEXT   'INITIALIZE 320 FOR ALB TEST'
       BYTE   0
*
IORGM  TEXT   'INITIALIZE 320 FOR ORIGINATE MODE'
       BYTE   0
*
IENBM  TEXT   'INITIALIZE 320 TO REENABLE RECEIVER'
       BYTE   0
*
ISQTM  TEXT   'INITIALIZE 320 TO SQUELCH RECEIVER'
       BYTE   0
*
ANSM   TEXT   'INITIALIZE 320 TO ANSWER MODE'
       BYTE   0
*
HONM   TEXT   'PUT LINE ON HOOK'
       BYTE   0
*
HOFFM  TEXT   'TAKE LINE OFF HOOK'
       BYTE   0
*
HELPM  TEXT   'TABLE OF COMMANDS'
       BYTE   >0D,>0A
       TEXT   'A ==> PUT MODEM IN ANSWER MODE'
       BYTE   >0D,>0A
       TEXT   'D ==> BLIND DIAL FOLLOWING DIGITS'
       BYTE   >0D,>0A
       TEXT   'E ==> ENABLE 320 RECEIVER'
       BYTE   >0D,>0A
       TEXT   'H ==> DISPLAY HELP LIST'
       BYTE   >0D,>0A
       TEXT   'J ==> PUT LINE ON HOOK'
       BYTE   >0D,>0A
       TEXT   'K ==> TAKE LINE OFF HOOK'
       BYTE   >0D,>0A
       TEXT   'L ==> RUN DIGITAL LOOP BACK TEST'
       BYTE   >0D,>0A
       TEXT   'M ==> RUN ANALOG LOOP BACK TEST'
       BYTE   >0D,>0A
       TEXT   'O ==> PUT MODEM IN ANSWER MODE'
       BYTE   >0D,>0A
       TEXT   'R ==> RUN THE 320 MODEM'
       BYTE   >0D,>0A
       TEXT   'S ==> SQUELCH THE 320 RECEIVER'
       BYTE   >0D,>0A
       TEXT   'Z ==> RESTART THE 7000'
       BYTE   >0D,>0A
       BYTE   0
*
HELLO  TEXT   'DSP MODEM. VERSION 1.0'
       BYTE   0
*******------------------------------------------*******
*****             command address table           *****
*******------------------------------------------*******
COMLIS DATA ANSMDM                  ; INITIALIZE TO ANSWER
       DATA ERR
       DATA ERR
       DATA DB                      ; dial command
       DATA ENBREC                  ; REENABLE RECEIVER ON 320
       DATA ERR
       DATA ERR
       DATA HELP                    ; HELP LIST
       DATA ERR
       DATA HOOKON                  ; TAKE LINE ON HOOK
       DATA HOOKOF                  ; TAKE LINE OFF HOOK
       DATA LDLB                    ; LOCAL DIGITAL LOOP BACK
       DATA IALB                    ; INITIALIZE TO ALB MODE
```

```
        DATA ERR
        DATA IORIG                  ; INITIALIZE TO ORIGINATE
        DATA ERR
        DATA ERR
        DATA B212                   ; RUN MODEM ROUTINE
        DATA SQTREC                 ; SQUELCH THE RECEIVER
        DATA ERR
        DATA ERR
        DATA ERR
        DATA ERR
        DATA ERR
        DATA ERR
        DATA INIT                   ; reset command
*******----------------------------------------*******
*****     INITIALIZE TO ALB MODE                 *****
*******----------------------------------------*******
IALB    EQU   $
        CLR   A             ; CYCLE THE CLEAR LINES
        STA   @MPRTC        ; OF THE I/O CONTROL
        ORP   %BIT0,CPORT   ; SET ALB INIT COMMAND
        ORP   %>F0,CPORT    ; PUT 320 IN INIT COMMAND MODE
IALB1   BTJOP %BIT6,DPORT,IALB1    ; CHECK 320 RESPONSE
        MOVD  %IALBM,MSGL       ; GET CONFIRMATION MESG
        CALL  @PRINT            ; AND SEND IT
        CLR   A                 ; CLEAR OUT THE COMMAND
        STA   @MPRTC            ; FROM I/O LINES
        BR    @TOP              ; EXIT ROUTINE
*******----------------------------------------*******
*****     INITIALIZE TO ORIGINATE MODE           *****
*******----------------------------------------*******
IORIG   EQU   $
        CLR   A             ; CYCLE THE CLEAR LINES
        STA   @MPRTC        ; OF THE I/O CONTROL
        ORP   %BIT1,CPORT   ; SET ORIG INIT COMMAND
        ORP   %>F0,CPORT    ; PUT 320 IN INIT COMMAND MODE
IORG1   BTJOP %BIT6,DPORT,IORG1    ; CHECK 320 RESPONSE
        MOVD  %IORGM,MSGL       ; GET CONFIRMATION MESG
        CALL  @PRINT            ; AND SEND IT
        CLR   A                 ; CLEAR OUT THE COMMAND
        STA   @MPRTC            ; FROM I/O LINES
        BR    @TOP              ; EXIT ROUTINE
*******----------------------------------------*******
*****     INITIALIZE TO RECEIVER SQUELCHED       *****
*******----------------------------------------*******
SQTREC  EQU   $
        CLR   A             ; CYCLE THE CLEAR LINES
        STA   @MPRTC        ; OF THE I/O CONTROL
        ORP   %3,CPORT      ; SET SQT INIT COMMAND
        ORP   %>F0,CPORT    ; PUT 320 IN INIT COMMAND MODE
ISQT1   BTJOP %BIT6,DPORT,ISQT1    ; CHECK 320 RESPONSE
        MOVD  %ISQTM,MSGL       ; GET CONFIRMATION MESG
        CALL  @PRINT            ; AND SEND IT
        CLR   A                 ; CLEAR OUT THE COMMAND
        STA   @MPRTC            ; FROM I/O LINES
        BR    @TOP              ; EXIT ROUTINE
*******----------------------------------------*******
*****     INITIALIZE TO REENABLE RECEIVER        *****
*******----------------------------------------*******
ENBREC  EQU   $
        CLR   A             ; CYCLE THE CLEAR LINES
        STA   @MPRTC        ; OF THE I/O CONTROL
        ORP   %4,CPORT      ; SET ENB INIT COMMAND
        ORP   %>F0,CPORT    ; PUT 320 IN INIT COMMAND MODE
IENB1   BTJOP %BIT6,DPORT,IENB1    ; CHECK 320 RESPONSE
        MOVD  %IENBM,MSGL       ; GET CONFIRMATION MESG
        CALL  @PRINT            ; AND SEND IT
        CLR   A                 ; CLEAR OUT THE COMMAND
```

```
       STA    @MPRTC              ; FROM I/O LINES
       BR     @TOP                ; EXIT ROUTINE
*******---------------------------------------*******
*****    INITIALIZE TO ANSWER MODE              *****
*******---------------------------------------*******
ANSMDM EQU    $
       CLR    A             ; CYCLE THE CLEAR LINES
       STA    @MPRTC        ; OF THE I/O CONTROL
       ORP    %5,CPORT      ; SET ANS INIT COMMAND
       ORP    %>F0,CPORT    ; PUT 320 IN INIT COMMAND MODE
IANS1  BTJOP %BIT6,DPORT,IANS1    ; CHECK 320 RESPONSE
       MOVD   %ANSM,MSGL          ; GET CONFIRMATION MESG
       CALL   @PRINT              ; AND SEND IT
       CLR    A                   ; CLEAR OUT THE COMMAND
       STA    @MPRTC              ; FROM I/O LINES
       BR     @TOP                ; EXIT ROUTINE
*******---------------------------------------*******
*****    PUT LINE ON HOOK                       *****
*******---------------------------------------*******
HOOKON EQU    $
       ANDP   %NOT7,APORT         ; PUT MODEM BACK ON HOOK
       MOVD   %HONM,MSGL          ; GET CONFIRMATION MESG
       CALL   @PRINT              ; AND SEND IT
       BR     @TOP                ; EXIT ROUTINE
*******---------------------------------------*******
*****    TAKE LINE OFF HOOK                     *****
*******---------------------------------------*******
HOOKOF EQU    $
       ORP    %BIT7,APORT         ; TAKE OFF HOOK
       MOVD   %HOFFM,MSGL         ; GET CONFIRMATION MESG
       CALL   @PRINT              ; AND SEND IT
       BR     @TOP                ; EXIT ROUTINE
*******---------------------------------------*******
*****    DISPLAY HELP LIST                      *****
*******---------------------------------------*******
HELP   EQU    $
       MOVD   %HELPM,MSGL         ; GET CONFIRMATION MESG
       CALL   @PRINT              ; AND SEND IT
       BR     @TOP                ; EXIT ROUTINE
*******---------------------------------------*******
*****              Clear command buffer         *****
*******---------------------------------------*******
CLEAR  CLR     A
       CLR     B
MORE   STA     @ADDBOT(B)          ; zero command register
       INC     B
       CMP     %40,B               ; are we done yet?
       JNE     MORE
       RETS
*******---------------------------------------*******
*****              Auto-answer routine          *****
*******---------------------------------------*******
INT1   BTJZP   %BIT1,APORT,ANSMOD ; DTR_ must be active, else return
       RETI
*
ANSMOD CLR     S4
       ORP     %BIT0,BPORT        ; Turn off CTS_
       MOVP    %>2A,IOCNT0        ; activate timer interrupt
       EINT
RIHIGH ORP     %BIT1,IOCNT0
       BTJOP   %BIT1,IOCNT0,RIHIGH ; Wait RI to fall
       ORP     %BIT1,IOCNT0
       MOVD    %50,COUNT1
STALOW MOVD    %10,MSTIME
       CALL    @MSDLY
       BTJOP   %BIT1,IOCNT0,RIHIGH ; separate rings
       DJNZ    COUNT1,STALOW
```

```
*
        MOVD    %RCALL,MSGL          ; send ring message
        CALL    @PRINT
        ORP     %BIT1,IOCNT0
*
LABEL0  INC     S4                   ; increment ring counter
        CMP     S3,S4
        JZ      PICKUP
NXTRNG  MOVD    %100,COUNT1
RILOW   MOVD    %100,MSTIME
        CALL    @MSDLY
        BTJOP   %BIT1,IOCNT0,RIHIGH ; check RI_ every 100 msecs
        DJNZ    COUNT1,RILOW
*
* no rings, abort answer
*
        ANDP    %NOT0,BPORT
        RETI
*
* Pickup the phone and go through answer procedures.
*
PICKUP  ORP     %BIT7,APORT          ; Go off hook
        ORP     %BIT1,BPORT          ; DSR is active
*
*  wait at least 2 seconds for billing delay
*
        MOV     %2,VALUE             ; must wait at least 2 secds
BDELAY  CALL    @SECDLY              ; Wait 2 seconds
        MOV     %18,CWT1             ; Initialize carrier abort timer.
        CLR     CWT2
        ORP     %BIT2,IOCNT0         ; Enable carrier abort interrupt
*
* determine if B212A or B103J mode
*
        ANDP    %BIT4,CPORT          ; answer mode (to 32010)
        ORP     %BIT4,APORT          ; ATE=1
        ANDP    %NOT2,APORT          ; Unsquelch 532, send 2225hz tone
*
        MOVD    %600,INT5TM          ; load timer
*
ORGWT0  BTJZP   %BIT5,DPORT,BE212 ; check for EDT_
        BTJZP   %BIT0,APORT,BE103 ; check for DCD_
        JMP     ORGWT0               ; keep looping till carrier timer aborts
BE212   MOVP    %>0C,IOCNT1          ; enable INT5
        JMP     GOTEDT               ; BELL 212 selected
BE103   MOVP    %>0C,IOCNT1          ; enable INT5
*
* Bell 103J selected
*
        MOV     %150,COUNT
DCDWT0  BTJOP   %BIT0,APORT,ORGWT0 ; check for DCD_
        MOVD    %1,MSTIME
        CALL    @MSDLY
        DJNZ    COUNT,DCDWT0
*
        MOVP    %>00,IOCNT0          ; Got DCD_, disable abort interrupt
        MOVP    %>00,IOCNT1
*
        MOVD    %CONN3,MSGL
        CALL    @PRINT
*
        MOVD    %765,MSTIME
        CALL    @MSDLY
        BR      @DAT103
*
GOTEDT  MOV     %150,COUNT           ; EDT_ active for at least 150 ms
EDTWT2  BTJOP   %BIT5,DPORT,ORGWT0
```

```
        MOVD    %1,MSTIME
        CALL    @MSDLY
        DJNZ    COUNT,EDTWT2
*
        MOVP    %>00,IOCNT0          ; Got EDT_, disable abort interrupt
        MOVP    %>00,IOCNT1
        ORP     %BIT2,APORT          ; Squelch 532
        ANDP    %NOT4,APORT          ; ATE=0 (EXI MODE)
        MOVD    %CONN12,MSGL
        CALL    @PRINT               ; CONNECT 1200
        ANDP    %NOT0,BPORT          ; CPORT is active (CTS_=0)
*
        MOVD    %765,MSTIME          ; Wait 765 ms
        CALL    @MSDLY
        BR      @B212                ; 212A mode, act as 32010 to DTE interface
*
*    Call Initiation Routines.
*
* We are now in data mode.  Wait for a disconnect.
*
DAT103 ANDP     %NOT0,BPORT          ; Activate CTS_
*
* look for escape character
*
LP103A MOV      %3,TEMP1
LP103B EQU      $
*        BTJOP   %BIT1,APORT,NODTR0 ; no DTR_
        BTJOP   %BIT0,APORT,DIS103 ; no DCD_
LP103E BTJZP    %BIT1,SCTL0,LP103E ; received char?
        MOVP    RXBUF,A
        CMP     S5,A                ; escape character?
        JNE     LP103A
        DJNZ    TEMP1,LP103B
*
* we now have three escape characters. start escape code timer
*
        MOV     %50,COUNT1
LP103C MOVD     %20,COUNT
LP103D MOVD     %1,MSTIME
        CALL    @MSDLY
        BTJOP   %BIT1,APORT,NODTR0 ; no DTR_
        BTJOP   %BIT0,APORT,DIS103 ; no DCD_
        BTJOP   %BIT1,SCTL0,LP103A
        DJNZ    COUNT,LP103D
        DJNZ    COUNT1,LP103C
*
* everything checked out O.K.
*
        JMP     CM103
*
NODTR0 MOV      %5,COUNT             ; 5 m/s check of DTR_
NODTR1 MOVD     %1,MSTIME
        CALL    @MSDLY
        BTJZP   %BIT1,APORT,LP103B
        DJNZ    COUNT,NODTR1
*
* Disconnect  from 103 data mode
*
DIS103 ORP      %BIT2,APORT          ; Squelch 532
        ANDP    %NOT7,APORT          ; Go on hook
        MOVP    %>03,IOCNT0          ; Enable interrupt 1
        MOVD    %NOCAR,MSGL          ; Send disconnect message
        CALL    @PRINT
TCODE2 BTJZP    %BIT0,SSTAT,TCODE2
        EINT
        BR      @INIT
*
```

```
*  103 COMMAND MODE
*
CM103   ANDP    %NOT0,BPORT          ; Activate CTS_
        MOVD    %RESET,MSGL
        CALL    @PRINT
        BR      @LOOK                ; look for new command
*
*************************************************************
****    TIMOUT INTERRUPT OF CARRIER DETECT               ****
*************************************************************
INT2    EQU     $
        DECD    CWT2                 ; DECREMENT SECONDARY COUNTER
        JNC     CABORT               ; IF COUNTED OUT THEN APORT
        RETI                         ; TIMOUT NOT COMPLETE CONTINUE
CABORT  ANDP    %NOT7,APORT          ; GO ON HOOK
        ORP     %BIT2,APORT          ; SQUELCH 532
        ANDP    %NOT0,BPORT          ; ACTIVATE CTS
        ORP     %BIT3,IOCNT0         ; DISBLE TIMER
        EINT
        MOVD    %NOCAR,MSGL          ; SEND NO CARRIER
        CALL    @PRINT               ; MESSAGE TO DTE
        BR      @LOOK                ; LOOK FOR NEXT COMMAND
*
INT3    RETI
INT4    RETI
INT5    RETI
        AORG    >FFF4
VECT5   DATA    INT5
VECT4   DATA    INT4
VECT3   DATA    INT3
VECT2   DATA    INT2
VECT1   DATA    INT1
VECT0   DATA    INIT
```