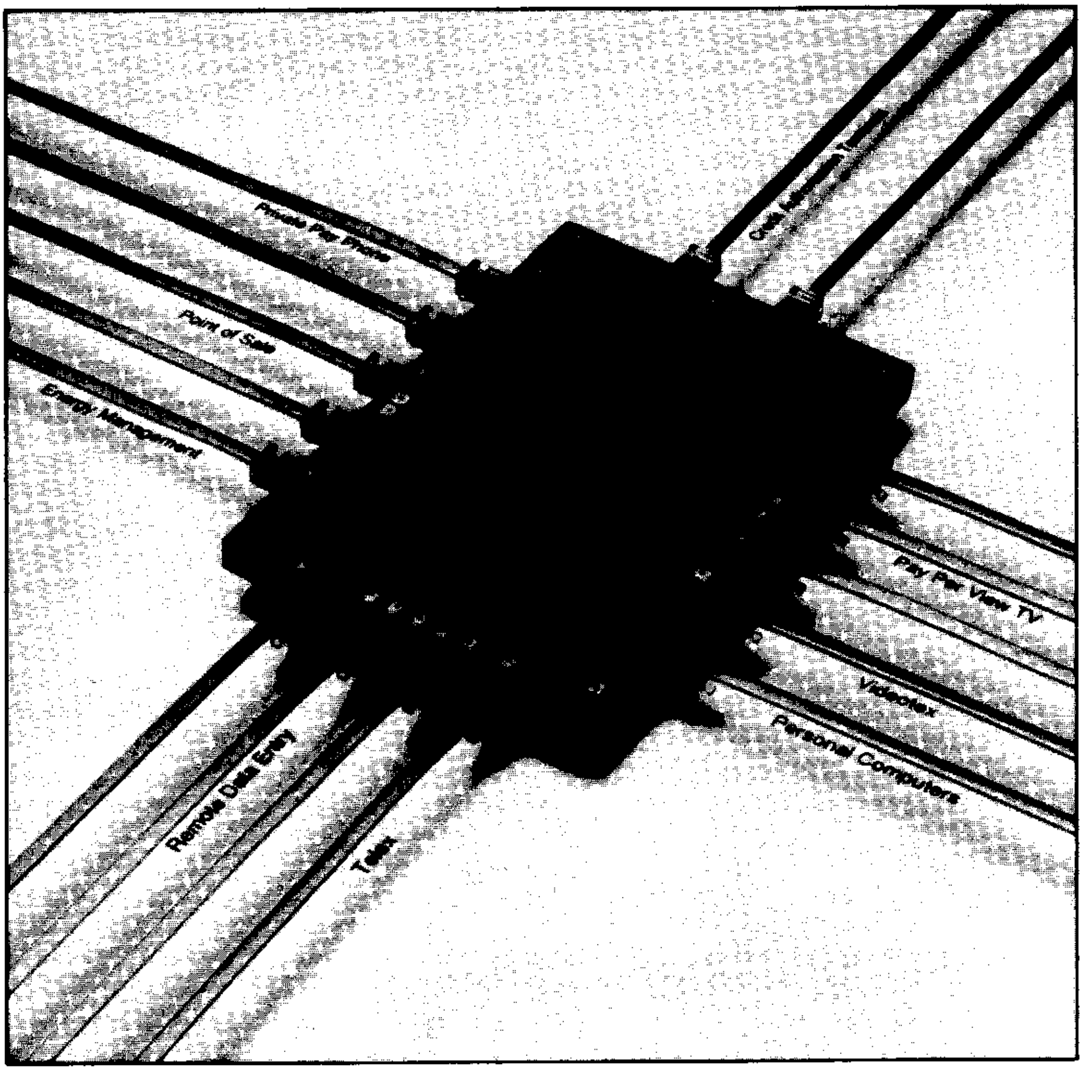
Modem

Technical Manual

Advanced
Micro
Devices

Am79101 Autodial FSK Modem WORLD-CHIP™
Am7910 FSK Modem WORLD-CHIP™
Am7911 FSK Modem WORLD-CHIP™



Advanced Micro Devices

MODEM TECHNICAL MANUAL

Am79101 WORLD-CHIP™ Autodial FSK Modem
Am7910 FSK WORLD-CHIP Modem
Am7911 FSK WORLD-CHIP Modem

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PREFACE

This manual provides technical descriptions and product application notes for Advanced Micro Devices' family of modem products. This manual is intended as a guide for modem designers; it also includes an introduction and sample designs for the newcomer to the modem field.

MANUAL ORGANIZATION

The Modem Technical Manual contains five chapters:

Chapter 1 – MODEM FAMILY – Introduces the AMD family of FSK modems and briefly describes their intended applications.

Chapter 2 – INTRODUCTION TO MODEMS – Provides a general overview of modem technology, a review of terminology and general concepts required to develop modem-related applications.

Chapter 3 – Am7910/11/101 FSK MODEMS – Describes the Am7910/11/101 product family as used in switched-network, leased-line and videotex applications.

Chapter 4 – DAA and HYBRID DESIGN – Describes the process of interfacing directly with the telephone network. It also explains the use of Hybrid Converters and Data Access Arrangements, and provides several sample designs built around the Am7910/11/101 FSK Modems.

Chapter 5 – PERFORMANCE TEST METHODS – Provides information regarding modem test setups and system performance measurements.

Appendix – Contains abbreviated Product Specifications for each product. The basic description, block diagram, pin-outs, and ordering information are shown. The detailed electrical specifications are contained in the complete Product Specifications available from the AMD sales offices or distributors.

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CHAPTER 1

CHAPTER 1
Modem Family

Advanced Micro Devices offers a family of high-performance FSK modem chips that may be easily integrated in system designs to interface terminals and workstations to the Telephone Network. The product family consists of:

- Am7910 FSK Modem
- Am7911 FSK Modem
- Am79101 Autodial FSK Modem

1.1 FSK MODEMS

AMD currently offers three single-chip modems in the Frequency Shift Keying (FSK) series including the Am7910, the Am7911 and the Am79101. This family of modems includes most of the building blocks required for a complete communication system. Some of the on-chip features include: analog-to-digital and digital-to-analog converters, internal crystal oscillator, and the essential RS-232/CCITT V.24 terminal control signals with TTL levels. A DAA (Data Access Arrangement) or acoustic coupler must be supplied externally to provide the Phone Line Interface.

The FSK modem chips all have the same basic structure as shown in the simplified block diagram in Figure 1.1. The Am7910 and the Am7911 differ mainly in timing parameters with the Am7911 being tailored for better performance in leased-line, multi-drop applications. The Am7911 also includes an extended set of modem selection modes.

The Am79101 has the Am7910's timing, the Am7911's extended mode set, plus autodial support, call progress tone detection and an internal hybrid.

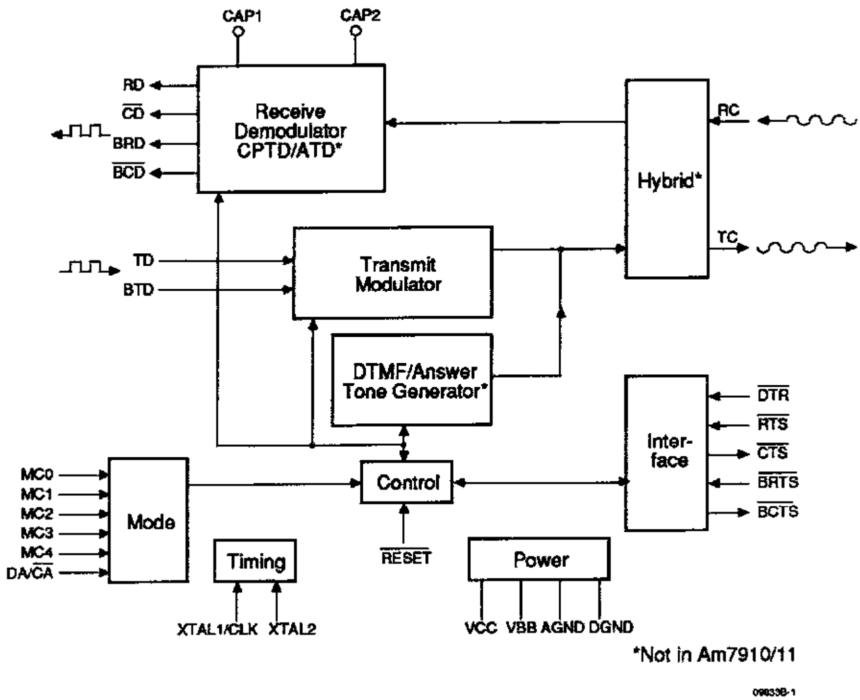


Figure 1.1 FSK Modems Basic Structure

The features of each member of the FSK Modem series are summarized below and compared in Table 1.1 See the respective Product Specification for complete details:

Am7910 Product Specification

Am7911 Product Specification

Am79101 Product Specification

1.1.1 Am7910 WORLD-CHIP™ FSK Modem

Developed for switched-network applications.

- Single-chip asynchronous FSK voice-band modem
- Pin-selectable data rates of 300, 600 or 1200 bps
- 300 bps Full-Duplex operation
- 1200 bps Full-Duplex Using a 4-Wire System
- 1200 bps Half-Duplex operation
- Compatible with Bell standards for 103/113/108/202
  - 5 bps Back Channel (B202)
- Compatible with CCITT standards for V.21/V.23
  - Up to 75 bps Back Channel (V.23)

Table 1.1 FSK Modem Family

Am7910/11/101 Common Capabilities

- Complete FSK Modem in 28-pin package
- 300 bps FULL-DUPLEX OPERATION
- 1200 bps HALF-DUPLEX OPERATION
- Compatible with Bell 103/113/108, Bell 202, CCITT V.21, and V.23
- Available in CDIP, PDIP and PLCC packages
- Commercial, Industrial and Extended temperature range
- No external filtering required
- All digital signal processing, digital filtering, and A/D–D/A conversion on-chip
- Includes essential RS-232/CCITT V.24 handshake signals on-chip

Unique Device Capabilities

Am79101	Am7910	Am7911
– Autodial Support	– Autoanswer capability	– Autoanswer capability
– DTMF Generation	– Dial-up network response times	– Fast response time for leased-line networks
– Call Progress Tone Detection	–1200 bps Full-Duplex on 4-wire	– 1200 bps Full-Duplex on 4-wire
– Answer Tone Detection	– Bell 202 with 5 bps back channel	– Bell 202 with 5 bps or 150 bps back channel
– On-Chip 4- to 2-Wire Hybrid	– V.23 with up to 75 bps back channel	– V.23 with up to 150 bps back channel
– Dial-up Network Response		
– Autoanswer Support		
– Bell 202 with 5 bps or 150 bps back channel		
– V.23 with up to 150 bps back channel		

09560A-T1.1

1.1.2 Am7911 WORLD-CHIP ™ FSK Modem

Developed primarily for leased-line, telex and videotex applications.

- Am7910 features plus enhancements
- Short RTS-CTS delay for leased-line multidrop applications
- Back Channel Performance
  - Bell 202, 5 bps or 150 bps
  - V.23, Up to 150 bps

1.1.3 Am79101 WORLD-CHIP Autodial FSK Modem

The first single-chip FSK modem to include on-chip autodial and autoanswer support.

- Extension of Am7910 features
- On-chip 4- to 2-wire hybrid conversion
- On-chip autodial support
  - DTMF tone generation
  - Call progress tone generation and detection
  - Answer tone generation and detection
- Dial-up network response times

CHAPTER 2

CHAPTER 2

Introduction To Modems

MODEM is an acronym formed from MOdulator/DEModulator. A modulator converts digital baseband data, normally in serial format, to an analog carrier for transmission over the telephone network. A demodulator receives a modulated analog carrier of similar nature and converts the analog signal to digital data. The digital data signal (Figure 2.1) is typically a random sequence of pulses which has a continuous power spectrum (Figure 2.2). This spectrum extends from zero to infinite frequency, hence transmission via a telephone channel is not possible because a phone line is band-limited to frequencies between 300 and 3400 Hz.

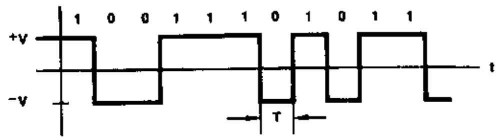


Figure 2.1 RS-232C Serial Digital Input Data

2.1 VOICEBAND MODEMS

Modems are generally characterized by speed and modulation techniques for which various standards exist (Table 2.1). Low speed modems (up to 1200bps) are implemented using frequency shift keying (FSK) modulation. Medium speed modems (1200 to 4800bps) are implemented using phase shift keying (PSK), or quadrature amplitude modulation (QAM). Higher speed modems (9600bps) are also implemented using quadrature amplitude modulation (QAM). PSK is really a special case of QAM.

These modulation techniques differ in the method of encoding data into an analog carrier (frequency, phase, amplitude), the number of bits encoded per modulation interval (baud), efficiency of transmission, frequency spectrum usage, and complexity of the circuitry required for implementation. The frequency bandwidth required by a modulation technique depends on the method of carrier data encoding and the rate at which this modulated carrier changes. This rate of change is the symbol interval or baud rate. A symbol may represent one or more bits.

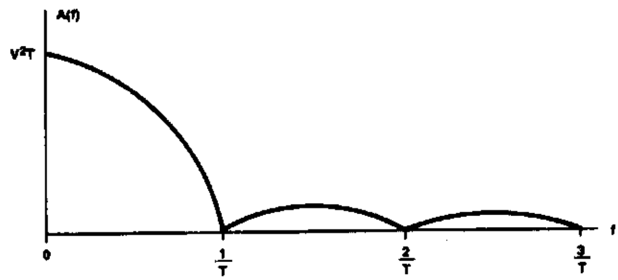
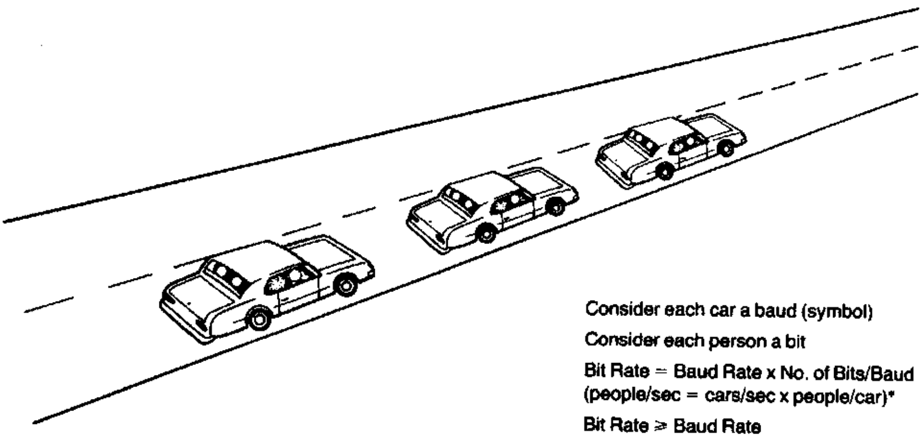


Figure 2.2 Power Spectrum of Digital Input Data

Table 2.1 Voiceband Modems

Speed	2-Wire Mode	Modulation	Specs
0–300bps	Full Duplex	FSK, Asynchronous	Bell 103/113 CCITT V.21
0–1200bps	Half Duplex (with Back Channel)	FSK, Asynchronous	Bell 202 CCITT V.23
1200bps	Full Duplex	DPSK, Synchronous	Bell 212A CCITT V.22 Vadic 3400
2400bps	Half Duplex (with Back Channel)	DPSK, Synchronous	Bell 201 CCITT V.26
2400bps	Full Duplex	QAM, Synchronous	CCITT V.22 bis
4800bps	Half Duplex	DPSK, Synchronous	Bell 208 CCITT V.27
9600bps	Half Duplex	QAM, Synchronous	Bell 209A CCITT V.29

Think of the information being transmitted as a stream of automobiles with people in them moving down a highway (the phone line).



*This analogy assumes that there are an equal number of people in each car.

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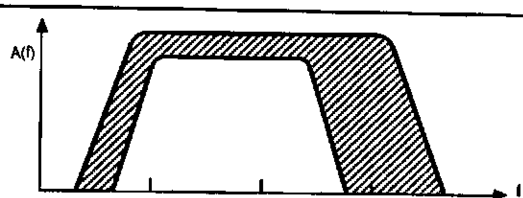
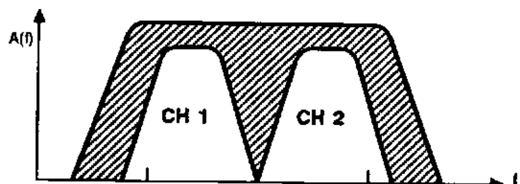
Figure 2.3 Bit Rate vs Baud Rate

As a simple analogy consider a stream of cars with people in them moving down a highway (Figure 2.3). The highway represents the phone line, the cars represent a baud, and the people represent bits. The bit rate equals the baud rate (number of cars per second) times the number of bits encoded per baud (number of people per car).

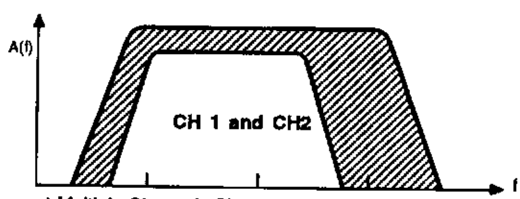
The phone line has a finite usable bandwidth which can be utilized by modems in various ways. A single high speed channel can use all the bandwidth (Figure 2.4a) or several lower speed channels can share the bandwidth by operating in different parts of the spectrum (Figure 2.4b). This latter method of sharing bandwidth is frequency division multiplexing (FDM).



a) Single Channel Uses More Than Half the Available Bandwidth (Bell 202, CCITT V.23)



b) Multiple Channels Share Bandwidth via FDM (Bell 103, CCITT V.21)



c) Multiple Channels Share Bandwidth via Echo Cancellation (CCITT V.32)

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Figure 2.4 Telephone Network Bandwidth Usage

2.2 FREQUENCY SHIFT KEYING (FSK)

Frequency shift keying is a modulation technique which encodes one bit of the serial data stream per baud. A logic 'one' in the bit stream places a mark frequency (f_M) on the phone line. A logic 'zero' places a space frequency (f_S) on the line. As the bit stream switches between one and zero, the analog signal on the line modulates between f_M and f_S (Figure 2.5). The modulation process generates energy over a broad spectrum, not only at the two frequencies, f_M and f_S. This spectrum depends on the sequence of bits in the serial data stream. An alternate one/zero pattern (1:1 or square wave) generates a simple line spectrum (Figure 2.6a). A 511-bit pseudo-random pattern of data bits (a good model of most data transferred via a modem) approximates a broad band spectrum (Figure 2.6b).

Since it encodes only one bit per baud, FSK uses approximately one Hz of bandwidth for each bit per second of data rate. For example, at a bit rate of 1200bps, a substantial portion of the phone line bandwidth is used, allowing only a single channel to be transmitted. At 300bps two independent channels can be accommodated within the line's bandwidth using FDM. Although it is the least efficient bandwidth user, FSK typically requires the least amount of hardware for its implementation.

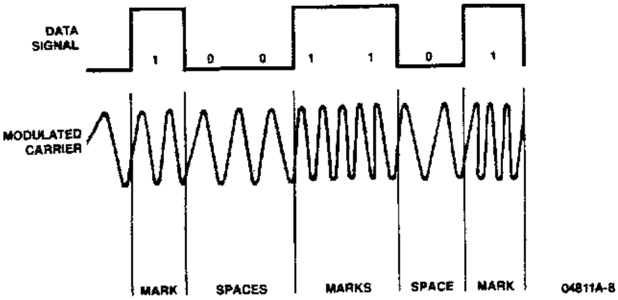
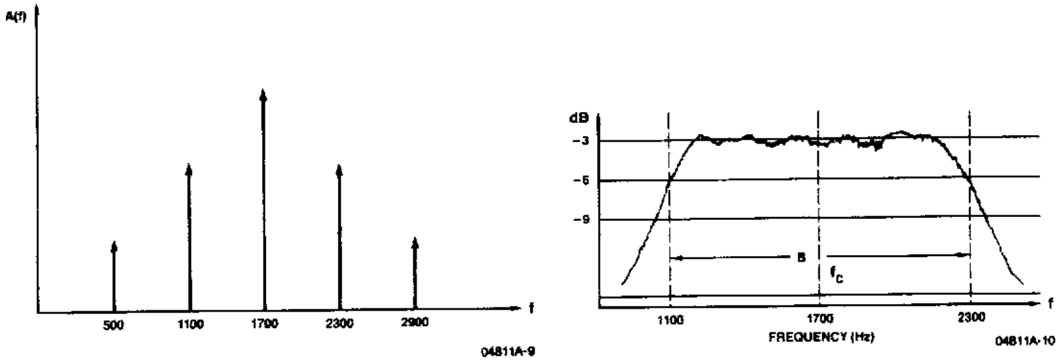


Figure 2.5 Modulated FSK Signal



a) 1 : 1 Pattern

b) 511-Bit Pseudo-Random Pattern

Figure 2.6 FSK Signal Spectrums

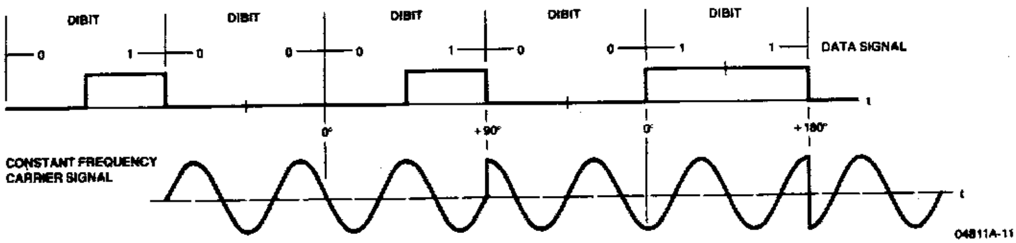
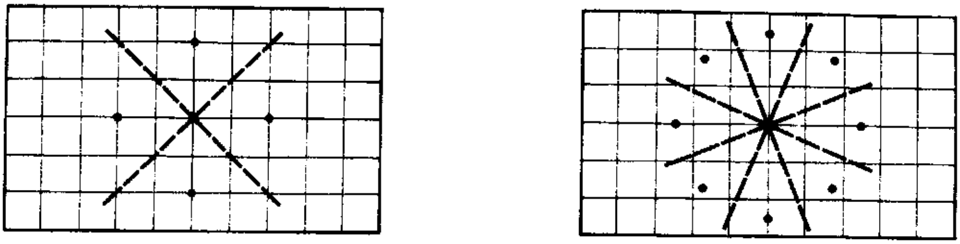


Figure 2.7 DPSK Modulation (4-phase angles)

2.3 PHASE SHIFT KEYING (PSK)

Phase shift keying is a modulation technique which encodes more than one bit of the serial data stream into a modulation symbol. Sequential bits in the data stream are grouped into pairs or triplets. When grouped into pairs for example, a two-bit code (dibit) is formed which selects one of four phase shifts to be applied to a carrier on the phone line. A three-bit code (tribit) selects one of eight phase shifts. Since spectrum usage is determined by the symbol or baud rate, encoding more bits per symbol allows a higher bit rate for a given bandwidth.

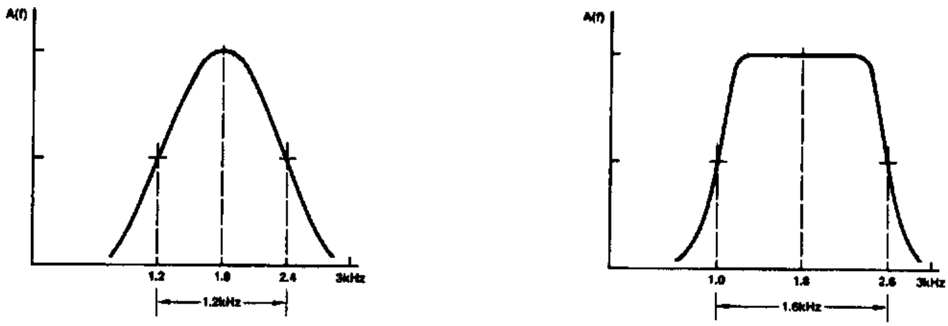
The encoding of two bits per baud and the corresponding phase shifted carrier is depicted in Figure 2.7. Phase shifts for particular modem specifications are generally indicated as point constellations (Figure 2.8). The constellation for the dibit encoding of Figure 2.7 is shown in Figure 2.8a. The power spectrums generated by these constellations are graphed in Figure 2.9.



a) 4-angle DPSK (2 bits/baud) 04811A-12

b) 8-angle DPSK (3-bits/baud) 04811A-13

Figure 2.8 DPSK Modulation Point Constellations



a) 4-angle DPSK (2400bps) 04811A-14

b) 8-angle DPSK (4800bps) 04811A-15

Figure 2.9 DPSK Power Spectrums

Spectrum use for two-bit PSK encoding allows two independent channels within the phone line bandwidth using FDM at a rate of 1200bps. A single channel at 2400bps is another way to use the available bandwidth with two-bit encoding. Three-bit coding permits a single channel at 4800bps.

The demodulation of PSK from the phase-shifted carrier into two or three-bit codes requires more sophisticated hardware than does FSK demodulation. This technique is more sensitive than FSK to phone line distortions, but these can be compensated for by more complex and costly processing in the modem. Higher data rates typically mean higher costs.

2.4 QUADRATURE AMPLITUDE MODULATION (QAM)

Quadrature amplitude modulation also encodes multiple data bits into a modulation symbol. In addition to phase shifting the carrier on the line, QAM modulates its amplitude. Typically, four sequential bits are encoded into a constellation such as shown in Figure 2.10. With this four-bit encoding, a single channel at 9600bps can be provided on the line. Figure 2.11 shows the power spectrum of a typical QAM signal.

QAM is even more sensitive to line distortions than PSK. Automatic equalizers are required to compensate for these distortions. This type of modem is the most complex and expensive of all the modems to implement.

2.5 ASYNCHRONOUS, SYNCHRONOUS TRANSMISSION

The serial data stream enters and exits a modem either asynchronously or synchronously. An asynchronous data stream may have a symbol rate (baud or bit) which varies from zero to the maximum permitted by the modulation technique. The bit rate is determined by the data and there is no separate clock signal to qualify the bits in the data stream. FSK modems are asynchronous.

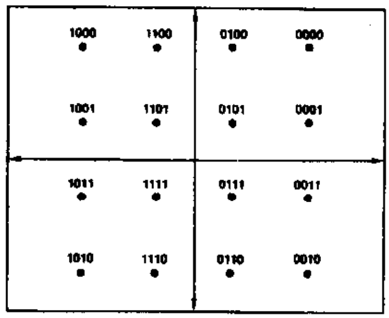


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Figure 2.10 QAM Point Constellation

A synchronous data stream has a fixed bit rate (± some small percent variation) determined by the modulation technique. Timing clocks are required to qualify the transmitted and received data streams. PSK and QAM modems are synchronous modulators, but asynchronous interface options are generally made available.

2.6 SWITCHED NETWORK vs LEASED LINES

The switched network is the medium used for most phone calls. The connection between two phones can be completely different each time a call is placed. Line characteristics such as amplitude and group delay for each connection vary depending upon physical length and particular routing of an individual call. In general, the amplitude and group delay distortion can be quite severe for a switched network connection. A switched network connection is a two-wire line; that is, a physical pair of wires that provides a single channel with a bandwidth of 300 to 3400Hz to carry information.

A leased or point-to-point line is a fixed connection between two phones. It always maintains approximately the same amplitude and group delay distortion characteristics. It may be conditioned by the phone company to have relatively flat amplitude and group delay responses. Typically, a lower bit error rate at a given bit rate can be obtained over a leased line than over a switched network line. Frequently, leased lines are four-wire lines; that is, two physical pairs of wires connect two phones. These four wires provide two independent channels of 300 to 3400Hz bandwidth to carry information.

2.7 TELEPHONE NETWORK BANDWIDTH USAGE

Simplex transmission is data transmission in only one direction during a given call (Figure 2.12). This is typically limited to two-wire lines.

Half-duplex transmission is data transmission bidirectional, but not simultaneous, over two-wire lines (Figure 2.13). Only one modem transmits and the other receives at any instant. Most of the available bandwidth can be used by this single transmission band. The process of reversing the direction of transmission is called "line turnaround." This is required if data must be transferred in both directions during a call. It is a time-consuming process, which can be avoided if most data is to be sent in one direction. For such cases, a small fraction of the bandwidth can be used to transmit data at a much lower rate in the direction opposite to, but simultaneous with, main channel transmission. This is called a backward or reverse channel. Half-duplex over two-wire lines is used at bit-rates from 1200 to 9600bps.

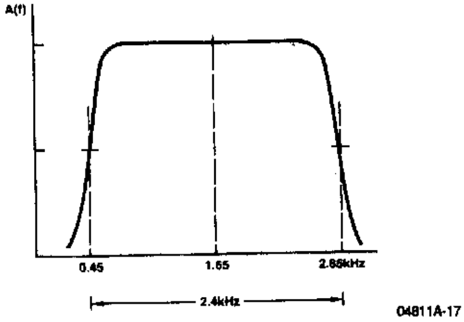


Figure 2.11 QAM Power Spectrum (9600bps)

Full-duplex transmission is simultaneous, bidirectional data transmission. Each modem can transmit and receive at the same time as shown in Figure 2.14. Available transmission bandwidth is divided into two channels via frequency division multiplexing (FDM) or the entire bandwidth is used for each channel via echo cancellation (Figure 2.4c). Low to medium data rates (300 - 2400bps) are possible over two-wire lines. Above 2400bps full-duplex operation usually requires four-wire lines. An advantage of full-duplex operation is that no line turnaround is required. Generally, the term full-duplex is qualified with phrases "over two-wire lines" or "over four-wire lines."

2.8 MODEM TERMINOLOGY

Acoustic Coupler
A device used for connecting the modem input and output to dial-up telephone lines by using the audio transmitter and receiver of a telephone handset.

Automatic Calling Unit (ACU)
A device used to dial telephone numbers automatically without manual user assistance.

Analog Loopback Mode
A diagnostic mode whereby the transmitter analog output (TC pin) is connected to the receiver analog input (RC pin). This test mode allows the local receiver to demodulate the data modulated by the local transmitter. Note that both the transmitter and receiver must be conditioned to work on the same frequency channel for this test mode to work. In normal operation, the transmitter and receiver work on different frequency channels.

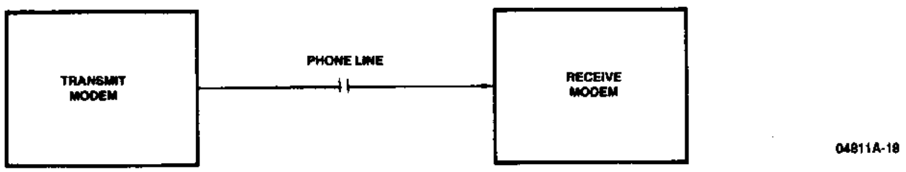


Figure 2.12 Simplex Data Transmission

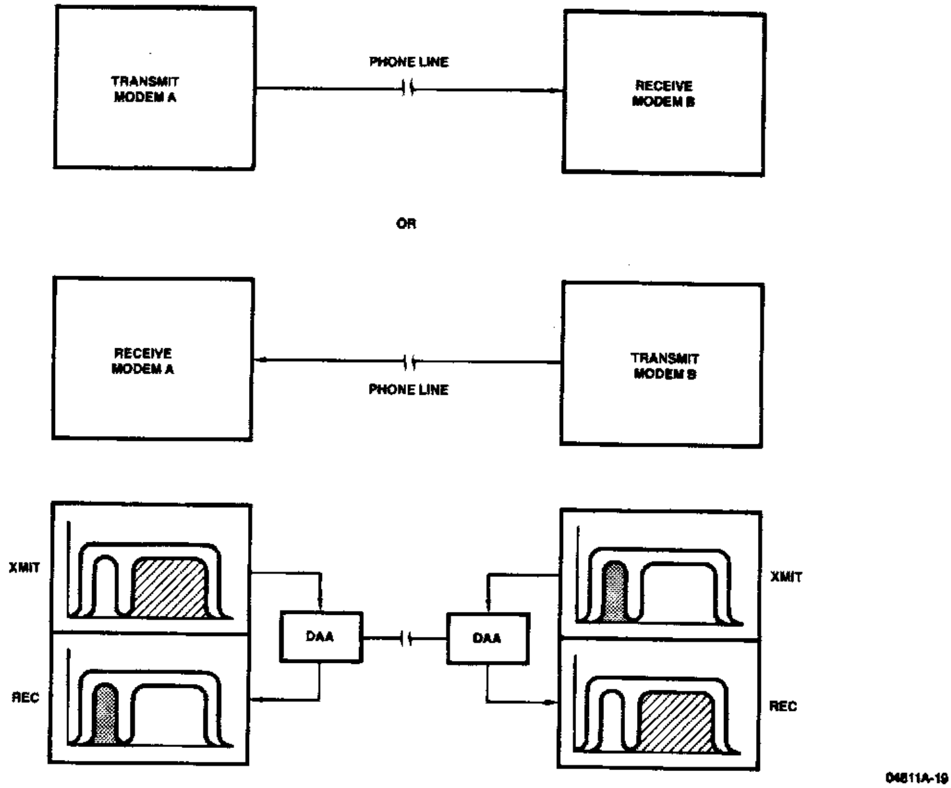


Figure 2.13 Half-Duplex Data Communication with Back Channel

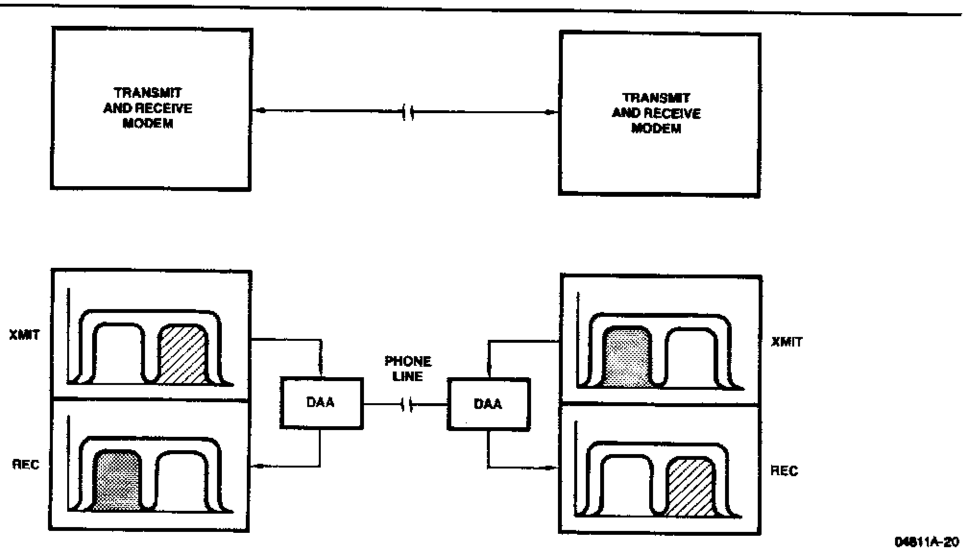


Figure 2.14 Full-Duplex Data Communication

Answer Tone
After automatically answering a call, the answering modem will transmit an answer tone back to the originating modem. This is an indication to the originating modem system that the remote modem has answered the call. In addition, answer tones are often times used to disable network echo suppressor devices.

Asynchronous Transmission
A data transmission scheme that handles data on a character-by-character basis without clock synchronization. The character code normally includes a "start" bit which signifies the beginning of a data character, 5 – 9 data bits, a parity bit and one, one and a half, or two stop bits.

Back Channel
A low data rate channel used on half-duplex modems to allow small amounts of data to be received or transmitted in the opposite direction of main channel data. In some cases, data may be transmitted on the back channel, while in others it is used only as an acknowledge channel.

Baseband
Information which exists in an unmodulated state. For instance, digital or analog data prior to modulation and transmission is baseband data.

Baud Rate
The number of modulation periods per second. For an FSK modem, one bit is encoded per modulation period; hence, the bit rate equals the baud rate. For higher speed modems, more than one bit is encoded per baud interval, so the bit rate equals the number of bits per baud times the number of bauds per second.

Bias Distortion
A measure of the variation from the ideal duration for a received mark and space. For a square wave (1:1) input pattern, bias distortion is a measure of the duty cycle of the output pattern. Bias distortion = duty cycle-50%. Bias distortion is data content independent.

Bit Error Rate (BER)
A measure of the average number of incorrectly demodulated bits by a modem receiver out of the total number of bits received.

Bit Rate (bps)
Number of bits transmitted per second.

Central Office (CO)
Common term describing the point at which subscriber telephone lines meet and are switched into other areas of the network. There are, however, a number of levels of central offices in the telephone network which provide different switching functions.

CCITT (Consultive Committee on International Telegraph and Telephone)
An international body concerned with the standardization of communications related specifications throughout the world.

Data Access Arrangement (DAA)
Protective circuitry required by the Federal Communications Commission (FCC) to protect the user from the sometimes harsh telephone network, and to protect the network from customer equipment malfunction. In the U.S., Part 68 of the FCC Rules discusses the requirements on connection to the network. A DAA is used for direct connection to the network – typically, through a modular telephone connector.

Data Terminal Equipment (DTE)
A term used in data communications to denote the source/destination of digital data. Common data terminal equipment devices are serial or parallel computer input/output ports and computer terminals.

dB (decibel)
A logarithmic ratio measurement of the amplitude or power of two signals. The equations for calculation of the number of dB's is shown below:

$$dB = 10 \times \log (P1/P2) \quad \text{(Power Ratio)}$$

or

$$dB = 20 \times \log (V1/V2) \quad \text{(Voltage Ratio)}$$

dBm
A logarithmic ratio measurement of the power of a signal relative to one milliwatt. Normally, the power of the signal is referenced to 600 ohms. The equation to calculate dBm is shown below:

$$dBm = 10 \times \log (P/0.001)$$

dBV
A logarithmic ratio measurement of the voltage of a signal relative to one volt. The equation to calculate dBV is shown below:

$$dBV = 20 \times \log (V)$$

Demodulator
The part of a modem which converts a modulated analog signal into a digital data output.

Digital Distortion
A form of distortion inherent to all data communications. Transitions which do not occur at the ideal transmission instants are digitally distorted. Isochronous, bias, jitter, peak individual and total peak distortion are types of digital distortion.

Digital Loopback
A test mode for phone line and remote modem quality determination. By connecting the digital receiver output (RD pin) to the digital transmitter input (TD pin) of a remote modem, modulated data can be transmitted to the remote modem and returned to the local modem. Since an analog medium, such as the PSTN, will be used to test the system, both the remote modem and the analog medium are under scrutiny. Note also that digital loopback is possible only for full-duplex modems, since two frequency channels must share the analog medium.

Direct Connection
Connection to the telephone network directly rather than using an acoustic coupler. Direct connection is regulated by various governmental bodies in most countries of the world.

Dual Tone Multi-Frequency (DTMF)
Telephone network signaling tones which consist of two distinct frequencies. Each of the frequencies used in DTMF is not harmonically related to any of the other DTMF frequencies. New or central offices use DTMF dial tones, and all touch-tone telephones generate DTMF signals when the telephone is being dialed.

Duplexor (Same as hybrid)
Circuit which matches the 4-wire modem signal (Transmit Carrier and GND, Receive Carrier and GND) with the two-wire telephone network.

Exchange (Same as Central Office)
Common term describing the point at which subscriber telephone lines meet and are switched into other areas of the network. There are, however, a number of levels of central offices in the telephone network which provide different switching functions.

Federal Communications Commission (FCC)
U.S. governmental agency which governs communications in the United States. Their work is more to set legal standards than to set technical standards. Part 68, which discusses direct connection to the switched telephone network, is a good example of the legal nature of the documents which the FCC publishes.

Frequency Shift Keying (FSK)
A frequency modulation technique whereby the modulated frequency is directly related to the digital bit or word present at the input to the modulator. For instance, a logical "1" at the modulator input will produce a certain frequency output while a logical "0" will produce a different frequency output.

Full-Duplex
Full-duplex data transmission allows simultaneous data transmission by two modems at the same speed in different directions.

Group Delay
The derivative of the phase of a signal with respect to frequency. A linear phase filter has constant group delay, a non-linear phase filter does not. Analog filters cannot have linear phase, whereas digital filters can be designed to have linear phase and hence constant group delay.

Half-Duplex
Half-duplex data transmission allows data transmission in only one direction at a time between two modems. The line may be "turned around" to allow data transmission in the other direction. In some cases, a "backward channel" is included to allow a low data rate channel in the opposite direction of the main data channel.

Handshaking
The protocols of the external modem control signals constitute the handshake. The control signals provided on a typical RS-232/V.24 connector for modem control are examples of handshake signals.

Hybrid, Two-to-Four-Wire
Circuit which matches the 4-wire modem signal (Transmit Carrier and GND, Receive Carrier and GND) with the two-wire telephone network.

Impulse Noise
Noise which enters the communications channel for a certain length of time and disappears. Typical impulse transients may be caused by lightning or central office switching. Impulse noise can often cause problems in the modem demodulation process.

Isochronous Distortion
A type of digital distortion which is a measure of the difference between the absolute earliest and the absolute latest transitions in a digital data stream. Transitions are compared to the ideal sample instants. Isochronous distortion is data content dependent.

Isochronous Transmission
In testing an asynchronous data communications device such as a modem, it is necessary for the test equipment to recognize the exact locations of transitions in the digital test patterns. Isochronous transmission sends data through an asynchronous test setup using synchronous test information.

Local Loop
The pair of telephone wires which exists from the customer's premises to the central office.

Local Loop Current
When the customer goes "off-hook", loop current flows from the central office to the customer through the local loop and back to the central office again. The central office senses the current flow as an indication that the customer desired to either dial or answer a call.

Mark
The frequency that corresponds to a logic "one" in frequency shift keying (FSK) modulation.

Modem
Acronym for MOdulator/DEModulator. A modem modulates baseband data for transmission over a communications channel and demodulates data from another modem over the same channel.

Modulator
A modem modulator changes baseband information, which is often times digital, into an analog carrier for transmission over a communications channel. The transmitter in a modem is a modulator.

Off-Hook
State in which the customer's equipment is electrically connected to the telephone network. When in the off-hook state, loop current flows from the central office through the user's equipment and back to the central office again.

On-Hook
State in which the customer's equipment is disconnected from the telephone network. When in the on-hook state, no loop current is able to flow from the central office because of an open relay at the customer's site.

Permissive Network Connection (U.S. only)
When connected to the telephone network permissively, the customer is required to limit his transmitted energy to a maximum of −9dBm.

Programmable Network Connection (U.S. only)
When a programmable network connection is used, the telephone company will install a programming resistor in the data communications equipment to set the transmit level of the customer's equipment from 0dBm to −12dBm in 1dB steps.

Progress Tones
Signaling tones provided by the central office to indicate network conditions to the customer. Examples of progress tones are: dial tone, ringback tone, and busy signal.

PTT
Post Telephone and Telegraph is the national governing body whose purpose is the regulation and administration of the telephone network.

Public Switched Telephone Network (PSTN)
The network available to standard users of telephone company equipment. Telephone lines in the home or office are connected to the PSTN when special data communications equipment is not used.

Signal-to-Noise Ratio (SNR)
The ratio of the desired energy level to an undesired energy level normally of a random nature and referred to as noise. SNR is typically given in decibel units.

Simplex
Simplex data transmission allows transmission in one direction only. The channel may not be turned around as in half-duplex modes.

Space
The frequency that corresponds to a logic "zero" in frequency shift keying (FSK) modulation.

CHAPTER 3

CHAPTER 3

Am7910, Am7911, Am79101 FSK Modems

Although the Am7910/11/101 FSK modems can be configured to operate with very few external components, the sophisticated user will want to incorporate them as a system block into a data communications system. This Chapter offers some details about each of the system components. A diagram of the data communications blocks required to implement a complete system using the Am7910/11/101 Modems is given in Figure 3.1.

3.1 DATA COMMUNICATIONS SYSTEM CONFIGURATION

In a typical application, a local computer terminal or CPU, has a requirement to transmit and receive data from a remote terminal or CPU using the telephone system as the transmission path. From the viewpoint of the computer or terminal, digital data communications and automatic phone dialing/answering functions are controlled by a Communications Control Unit (CCU). If the automatic phone dialing and answering functions (see Automatic Calling Functions) are not desired by the user, then a Universal Asynchronous Receiver/Transmitter (UART) can be used to control the digital data flow between the terminal and the modem. The UART type of control function is described more thoroughly in the Data Terminal Equipment (DTE) interface section.

After the digital data from the computer is conditioned by the communications controller, the modem handles conversion of the the digital data to an analog carrier (modulation). The demodulation process (analog carrier to digital data) is also controlled by the modem. Considerations for proper operation of the Am7910/11/101 single-chip FSK modems are presented below. The equipment connected to the telephone network must be protected from the external environment (lightning, power lines, etc.). In addition, the network must be

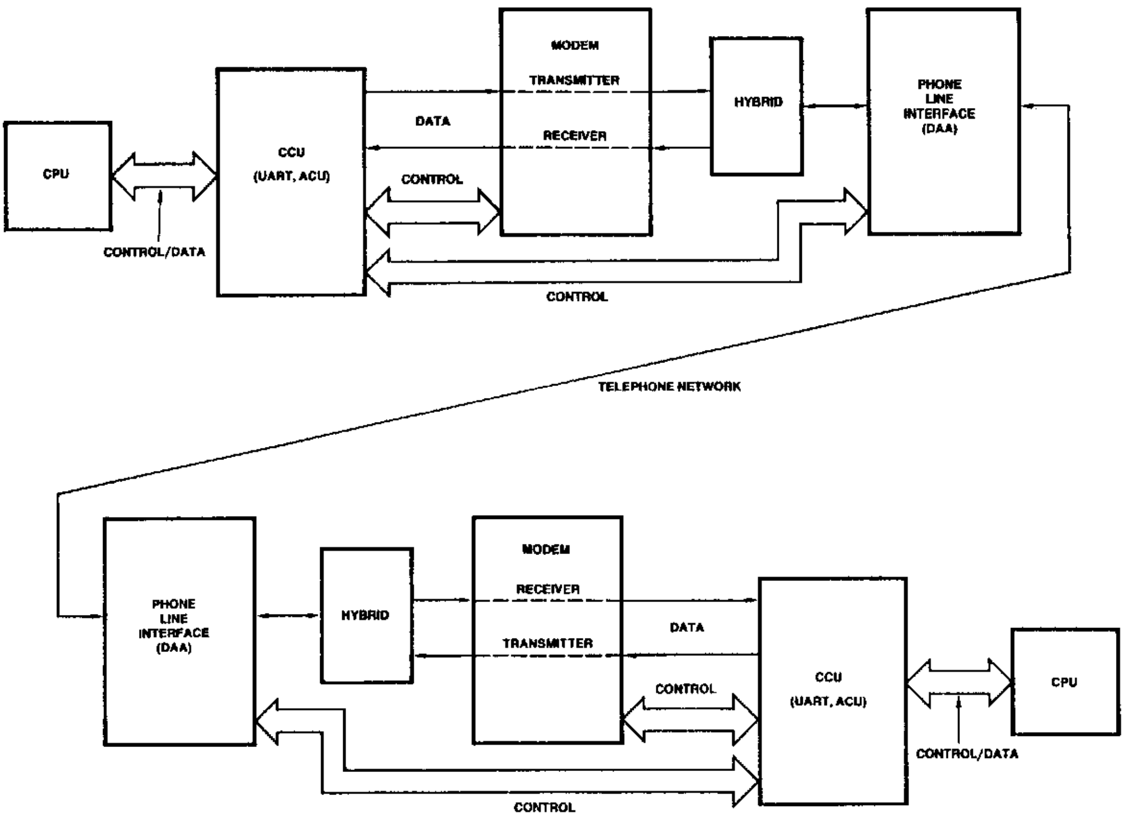


Figure 3.1 Data Communications System Block Diagram

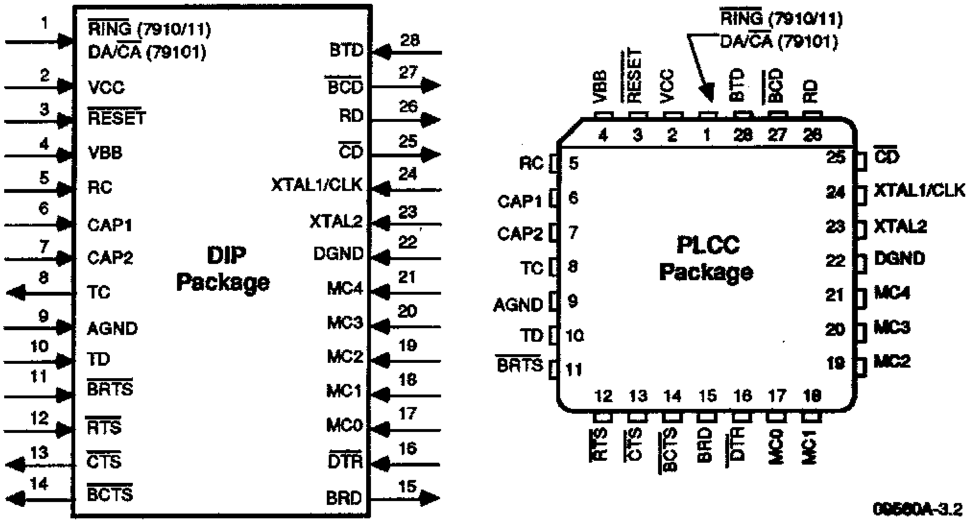


Figure 3.2 Am7910/7911/79101 FSK Modem Package Configurations

protected from improper connection of equipment. These tasks are assigned to a device known as a Data Access Arrangement (DAA).

Another method of network interface is known as acoustic coupling. Using a standard telephone (which already has the aforementioned protection) and a pair of acoustic transducers in a cradle, phone line connection can be accomplished. Phone line connection is described in more detail in the Phone Line Interface section 3.3.

3.2 EXTERNAL CONNECTIONS

Figure 3.2 shows the FSK modem family packaging configurations and external signal pins. The Am7910 and Am7911 are completely pin-for-pin compatible except for the CAP1/CAP2 resistor value which selects the RC time constant used by the analog-to-digital converter in the receiver section (described later). The Am79101 differs from the Am7910/11 regarding the definition of pin 1 and the carrier signal levels on pins 5 and 8.

3.2.1 Clock

The FSK modems use digital signal processing to perform both modulation and demodulation operations. Since the performance of the digital signal processor is directly dependent on the sampling (clock) frequency, the exact clock value of 2.4576 MHz with a tolerance of 0.01% is required for proper operation. Master timing of the modem is provided by either a crystal connected to the XTAL1 and XTAL2 inputs or an external clock applied to the XTAL1 input.

3.2.2 Crystal

When a crystal is used it should be connected as shown in Figure 3.3. The crystal should be a parallel resonance type (HC-33 holder recommended), and its value must be 2.4576 MHz ±0.01%.

3.2.3 External Clock

The external clock signal could be derived from a crystal-driven baud rate generator. It should be connected to the XTAL1 input with the XTAL2 input left floating. The timing parameters required of this clock are shown in Figure 3.3.

NOTE: Do not drive XTAL1 with a TTL driver and a pull-up resistor. The recommended external clock source is a CMOS driver (VCC = 5V).

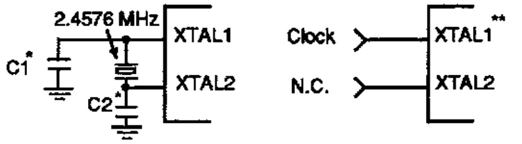
Carrier frequencies may be altered by using an improper clock frequency for the modem. The shift in the carrier frequency will be directly proportional to the shift in clock frequency. Note that the modem will no longer meet the Bell and CCITT standards when a shift in clock frequency occurs.

3.2.4 Reset

A RESET pin is provided on the modem which halts the processing of transmitter and receiver inputs. RESET does not control the handshake state machines. DTR should be brought High before any RESET pulse is applied. A High-to-Low transition on DTR is the only way to return the modem to a known state in the handshake state machine. DTR must not be tied Low.

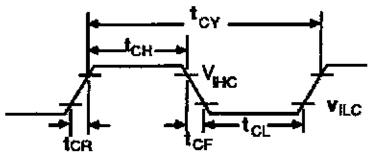
Manufacturer	P/N	C1	C2
M-Tron	MP-2	10pF	20pF
Monitor Products	MM-33	10pF	20pF

Note: Use crystal size HC-33 instead of HC-18



* Capacitor values vary with different crystal manufacturers

** The input impedance of the pin appears as 5-10 pF to ground in parallel with at least 1 MΩ resistance



Symbol	Parameters	Min	Typical	Max	Units
t_{CY}	CLK Period	406.86	406.9	406.94	ns
t_{CH}	CLK High Time	165			ns
t_{CL}	CLK Low Time	165			ns
t_{CR}	CLK Rise Time			20	ns
t_{CF}	CLK Fall Time			20	ns
V_{IHC}	CLK High Level	3.8		V_{CC}	V
V_{ILC}	CLK Low Level	-0.5		0.8	V

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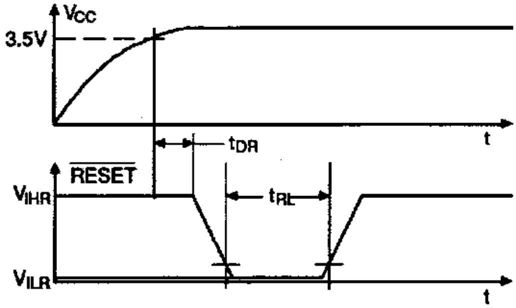
Figure 3.3 Clock Generation Specifications

While the Am7910/11/101 FSK modems have internal finite state machines with no unknown states, the one requirement for proper operation is that initial conditions be met after the system state has been corrupted (at power-on, system re-boot, etc.). Returning to initial conditions is done with a High on $\overline{DTR}$ followed by a Low pulse on $\overline{RESET}$.

The modem can be forced into the internal reset sequence by applying a Low level to the $\overline{RESET}$ input for at least one clock period (see Figure 3.4).



t_{DR} = delay from the time V_{CC} reaches 3.5 V and the falling edge of $\overline{RESET}$ signal (>1 µs)

t_{RL} = $\overline{RESET}$ Low duration time > 407 ns

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Figure 3.4 Reset Signal Specifications

3.2.5 Automatic Power-on Reset

Figure 3.5 shows a suggested circuit to initiate the internal reset sequence automatically whenever VCC is applied. VCC rise time should be faster than one half the RC time constant. The modem contains a diode to discharge the capacitor when VCC = 0V.

Note: VCC must be >3.5 V for at least 1.0 µs for proper reset operation. Between repetitive reset pulses, $\overline{RESET}$ must be High for at least 1.0 µs.

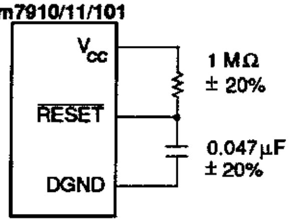

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Figure 3.5 Automatic Reset Circuit

3.2.6 CAP1 and CAP2 Connections

A resistor connected in series with a capacitor between pins CAP1 and CAP2 of the modems is required to select the RC time constant used by the analog-to-digital converter in the receiver. The values listed below will allow optimal A/D converter performance and in turn reduce the distortion of the receiver. A 10 % tolerance is allowed to eliminate the need for precision components.

Part Number	R value	C value
Am7910	100 Ω	2000 pF
Am7911	910 Ω	2000 pF
Am79101	910 Ω	2000 pF

The Am7910 and Am7911 are completely pin-for-pin compatible except for the CAP1/CAP2 resistor value.

3.2.7 Grounding

Analog ground (AGND) is a low-impedance ground for noise-sensitive analog circuitry and should be as "quiet" as possible.

Digital ground (DGND) is a relatively noise-tolerant ground and can be shared with other digital circuits on the same board.

No more than 50 mV offset should exist between AGND and DGND, and it is recommended that the two grounds be tied together at the power-supply common to avoid ground loops. Good grounding and power-supply practices are especially important if the modem chip is in a noisy environment, e.g., near RF circuitry or a switching power supply. Proper use of decoupling capacitors and bus terminators can reduce ground noise dramatically.

3.2.8 Power Supply

The modems have symmetric positive and negative power supplies (+5.0 V and -5.0 V). Symmetric supplies are required to provide power to the analog sections. The A/D convertor and D/A convertor are sensitive to ripple on either power supply.

As described in the Am7910/11/101 Product Specifications, power supply dc tolerance is ±5%. The Vcc rise time must be a minimum of 5.0 ms to ensure proper oscillator start-up. Power supply ripple must be less than 50 mV.

Decoupling VCC to DGND and VBB to AGND with 0.01 µF ceramic capacitors, connected as close to the device as possible, is recommended. It is important to note that excessive ripple will not only degrade performance, but it may also cause problems that are not obviously related to power supply disturbances.

3.3 PHONE LINE INTERFACE

The Am7910/11's analog output is Transmit Carrier (TC) and the analog input is Receive Carrier (RC). These two signals must be converted into one signal for connection to the phone line. A duplexor or hybrid (2-wire to 4-wire convertor) is used to allow the modem's signals to be attached to the telephone network (see Figure 3.6). The single output of the hybrid is connected to a Data Access Arrangement (DAA) which is a device physically connected to the phone line.

The Am79101 contains an on-chip hybrid which simplifies the DAA design (see Figure 3.7). Chapter 4 contains a detailed description of the hybrid for the Am7910/7911 and a description of a DAA.

3.3.1 Am7910/11 Analog Offsets

The maximum dc offset generated by the TC output is ±100 mV. Therefore, a blocking capacitor between TC and the hybrid is needed (Figure 3.8).

The simple method of meeting the ±30 mV max dc offset into the Am7910/11's RC pin, is to ac couple the RC input. A common pitfall is to isolate RC from the hybrid circuitry with a capacitor, without providing a path for dc bias current into RC (the RC input feeds an op amp). In this case, the auto-zero circuit of the modem overflows and the receive data pin will be set High or Low until the dc offset returns to within the specified limit. The recommended approach is to place a 100 kΩ resistor from RC to AGND with a 0.033 µF capacitor between RC and the hybrid. If alternate values are chosen, care should be taken to ensure that the received signal is not filtered.

3.3.2 Am7910/11 Transmit Levels

The transmit levels out of the Am7910/7911 are nominally -3 dBm into 600 Ω. The hybrid requires a minimum 6 dB loss from the Transmit Carrier Pin to the Telephone line to meet FCC Part 68.

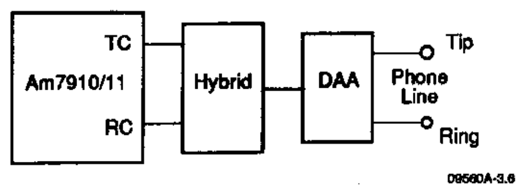

Figure 3.6 Phone Line Interface, Simplified

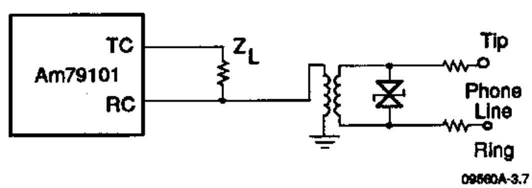

Figure 3.7 Data Access Arrangement

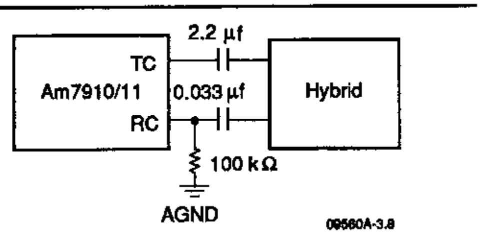

Figure 3.8 Hybrid 4-Wire to 2-Wire Converter

Table 3.1 Am79101 Nominal Transmit Levels

		FSK	DTMF (LOW)	DTMF (HIGH)
TC	dBm (600)	-3.0	-1.5	+0.5
	dBm (1200)	-6.0	-4.5	-2.5
	rms	0.548	0.652	0.820
	peak	0.776	0.922	1.160
RC	dBm (600)	-9.0	-7.5	-5.5
	dBm (1200)	-12.0	-10.5	-8.5
	rms	0.275	0.327	0.411
	peak	0.389	0.462	1.160
LINE	dBm (600)	-10.5	-9.0	-7.0
	dBm (1200)	-13.5	-12.0	-10.0
	rms	0.231	0.275	0.346
	peak	0.327	0.389	0.489

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3.3.3 Am79101 Transmit Levels

Since the Am79101 has a built-in hybrid, the resistor Z_L and the line transformer together give a 6 dB loss between TC and the transformer input (Figure 3.7). Table 3.1 provides levels at the TC pin, the transformer input (or RC) and levels on the line for FSK and DTMF.

3.4 INTERFACE DESCRIPTION

3.4.1 DA/$\overline{\text{CA}}$ (Am79101 only)

The Data/$\overline{\text{Call}}$ pin selects the Data mode (DA/$\overline{\text{CA}}$ = High) or Call mode (DA/$\overline{\text{CA}}$ = Low). In the Data mode, the Am79101 operates as a modem using the mode-control pins (MC4-MC0) to set the modem mode. In the Call mode, the mode control pins and $\overline{\text{RTS}}$ determine the generation of either DTMF or answer tones and the detection of call progress tones or answer tone.

The initial state and defaults of DA/$\overline{\text{CA}}$ should be the Data mode (DA/$\overline{\text{CA}}$ = High). To enter the Call mode, DA/$\overline{\text{CA}}$ and $\overline{\text{DTR}}$ must be High and a valid modem mode must be specified on the MC pins (i.e., no reserved modes). Then, when DA/$\overline{\text{CA}}$ is changed to a Low, the Am79101 will be in the Call mode. To return to the Data mode, DA/$\overline{\text{CA}}$ is taken High while there is a valid modem mode on the MC pins.

3.4.2 $\overline{\text{Ring}}$ (Am7910/11 Only)

This input signal permits autoanswer capability by responding to a ringing signal from a Data Access Arrangement. If a ringing signal is detected ($\overline{\text{RING}}$ = Low) and $\overline{\text{DTR}}$ is Low, the modem begins a sequence to generate an answer tone at the TC output.

Note: The Am79101 does not have a ring input pin. Rather, a ring indication output from the DAA interrupts the local microprocessor, which then initiates the autoanswer sequence in the Am79101, as described in section 3.8.

3.4.3 Mode Controls (MC0-MC4)

The FSK modem family has multiple built-in modem modes selectable by the user through a set of Mode Control Pins. Table 3.2 lists the modem modes, mode-control pin states and the product containing a particular mode.

The loopback modes set the receiver channel signal processing band to that of the transmit channel. No internal connection is made. The user must connect the TC pin to the RC pin if analog loopback is required (see Figure 3.9).

For digital loopback, external connection of the RD and TD pins is required.

With the Am7910/11, loopback modes can also be used to achieve full-duplex, 1200 bps communication. In CCITT V.23 or Bell 202 loopback modes, the modem can transmit and receive at 1200 bps using a 4-wire configuration (transmit over one channel and receive on another). See the System Configuration section for details.

3.4.4 $\overline{\text{Data Terminal Ready (DTR)}}$

A Low level on this input indicates the data terminal is ready to send and/or receive data via the modem. This signal is gated with all other TTL inputs and outputs so that a Low level enables these signals as well as the internal control logic. A High disables all TTL I/O pins and the internal logic.

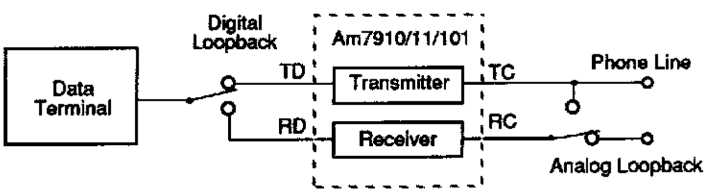

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Figure 3.9 Analog Loopback

Table 3.2a Am7910/11/101 Mode Control Lines

7910	7911	79101	DA/$\overline{CA}$	MC4	MC3	MC2	MC1	MC0	DESCRIPTION
X	X	X	1	0	0	0	0	0	Bell 103 Originate 300 bps Full-Duplex
X	X	X	1	0	0	0	0	1	Bell 103 Answer 300 bps Full-Duplex
X	X	X	1	0	0	0	1	0	Bell 202 1200 bps Half-Duplex
X	X	X	1	0	0	0	1	1	Bell 202 with equalizer
X	X	X	1	0	0	1	0	0	CCITT V.21 Orig 300 bps Full-Duplex
X	X	X	1	0	0	1	0	1	CCITT V.21 Ans 300 bps Full-Duplex
X	X	X	1	0	0	1	1	0	CCITT V.23 M2 1200 bps Half-Duplex
X	X	X	1	0	0	1	1	1	CCITT V.23 M2 with Equalizer
X			1	0	1	0	0	0	CCITT V.23 M1 (1) 600 bps Half-Duplex
	X	X	1	0	1	0	0	0	CCITT V.23 M1 (2) 600 bps Half-Duplex
			1	0	1	0	0	1	Reserved
	X	X	1	0	1	0	1	0	Bell 202 with 150 bps Back Channel
	X	X	1	0	1	0	1	1	Bell 202 with 150 bps Back & Equalizer
	X	X	1	0	1	1	0	0	CCITT V.23 M1 (2) with Soft Turn-Off (STO)
			1	0	1	1	0	1	Reserved
	X	X	1	0	1	1	1	0	CCITT V.23 M2 (2) with STO
	X	X	1	0	1	1	1	1	CCITT V.23 M2 (2) with STO and Equalizer
X	X	X	1	1	0	0	0	0	Bell 103 Orig. Loopback
X	X	X	1	1	0	0	0	1	Bell 103 Answer Loopback
X	X	X	1	1	0	0	1	0	Bell 202 Main Loopback
X	X	X	1	1	0	0	1	1	Bell 202 with Equalizer Loopback
X	X	X	1	1	0	1	0	0	CCITT V.21 Orig. Loopback
X	X	X	1	1	0	1	0	1	CCITT V.21 Ans. Loopback
X	X	X	1	1	0	1	1	0	CCITT V.23 M2 Main Loopback
X	X	X	1	1	0	1	1	1	CCITT V.23 M2 with Equalizer Loopback
X	X	X	1	1	1	0	0	0	CCITT V.23 M1 Main Loopback
X			1	1	1	0	0	1	CCITT V.23 (1) Back Loopback
	X	X	1	1	1	0	0	1	CCITT V.23 (2) Back Loopback
	X	X	1	1	1	0	1	0	Bell 202 (2) Back Loopback

(1) Up to 75 Baud Back Channel
(2) Up to 150 Baud Back Channel

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Table 3.2b Am7910/11/101 Mode Control Lines (Continued)

7910	7911	79101	DA/$\overline{CA}$	MC4	MC3	MC2	MC1	MC0	DESCRIPTION
			1	1	1	0	1	1	Reserved
			1	1	1	1	0	0	Reserved
			1	1	1	1	0	1	Reserved
			1	1	1	1	1	0	Reserved
			1	1	1	1	1	1	Reserved
		X	0	0	0	0	0	0	DTMF 0 and Answer Tone Detection
		X	0	0	0	0	0	1	DTMF 1
		X	0	0	0	0	1	0	DTMF 2
		X	0	0	0	0	1	1	DTMF 3
		X	0	0	0	1	0	0	DTMF 4
		X	0	0	0	1	0	1	DTMF 5
		X	0	0	0	1	1	0	DTMF 6
		X	0	0	0	1	1	1	DTMF 7
		X	0	0	1	0	0	0	DTMF 8
		X	0	0	1	0	0	1	DTMF 9
		X	0	0	1	0	1	0	DTMF *
		X	0	0	1	0	1	1	DTMF #
		X	0	0	1	1	0	0	Bell 103 Answer Tone
		X	0	0	1	1	0	1	Bell 202 Answer Tone
		X	0	0	1	1	1	0	V.21 or V.23 Answer Tone
		X	0	0	1	1	1	1	Call Progress Tone Detection
		X	0	1	X	X	X	X	Reserved

NOTE: Reserved modes should not be entered.
DA/$\overline{CA}$ applies to the AM79101 only.

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When $\overline{DTR}$ is High, the modem handshake state machine is reset to initial conditions. This is the only way to reset the state machine and must be done after power-up. The state machine does not automatically power up to a known state. If $\overline{DTR}$ is permanently enabled (Low), the state machine will simply run from wherever it powers up. This can result in abnormal behavior such as an unusually short $\overline{RTS}$-$\overline{CTS}$ delay due to lack of $\overline{DTR}$ initialization.

In order to change the modem mode while the modem is powered up, use the following sequence:

1. Take $\overline{DTR}$ High
2. Change mode inputs to desired configuration
3. Wait at least 100 µs
4. Take $\overline{DTR}$ Low

The mode inputs perform some hardware functions, and they are also sampled periodically by the state machine. If the mode inputs are changed without the re-initialization using $\overline{DTR}$, the state machine will not completely change to the new mode.

NOTE for the Am79101: In call mode, $\overline{DTR}$ does not have to be changed when going from one DTMF digit to another.

3.4.5 Request To Send (RTS)

A Low on this input instructs the modem to enter the transmit mode. This input must remain Low for the duration of data transmission. This signal has no effect if DTR is set High (disabled). A High level on this input turns off the transmitter.

For the Am79101 in the Call mode, RTS also controls the transmitter. When RTS is Low, the DMTF or answer tone specified by the MC pins will be transmitted. A High level will turn off the tone. During answer tone detection and call progress tone detection, RTS should be High.

3.4.6 Back Request To Send (BRTS)

Since the 1200 bps modem configurations (Bell 202 and CCITT V.23) permit only half-duplex operation over 2-wire lines, a low baud rate backward/back channel is provided for simultaneous transmission in the reverse direction. BRTS is equivalent to REQUEST TO SEND for the main channel, except that it belongs to the back channel. Since the modem contains a single transmitter, RTS and BRTS should not be asserted simultaneously. BRTS is meaningful only when a 202 or V.23 mode is selected by MC0-MC4. In all other modes, it is ignored.

For the V.23 modes and the 202 150 bps (or 75 bps) back channel mode, the frequency appearing at the transmitted carrier (TC) output pin is determined by a mark or space at the Back Transmitted Data (BTD) input.

For the 202 5 bps Back Channel Mode, a frequency of 387 Hz appears at TC when BRTS is Low and BTD is High. No energy (0.0 volts) appears at TC when BRTS is High. BTD should be fixed High for 202 back channel transmission. BRTS then, is equivalent to the transmitted data. BRTS is the Secondary Request-to-Send for 202 S/T modems, or the Supervisory Transmitted Data for 202 C/D modems.

3.4.7 Clear To Send (CTS)

This output goes Low at the end of a delay (t_{RCON}) initiated when RTS goes Low. Actual data to be transmitted should not be presented to the Transmit Data input until a Low is indicated on the CTS output. This gives the receiving modem (on the other end of the phone line) enough time to recognize a valid carrier signal before data is transmitted. Normally the user should force the TD input High whenever CTS is High so a mark will be sent during the t_{RCON} time. CTS goes High at the end of a delay initiated when RTS goes High (t_{RCOFF}). CTS will never be Low when DTR is High.

3.4.8 Back Clear To Send (BCTS)

This line is equivalent to CLEAR TO SEND for the main channel, except it belongs to the back channel. BCTS is meaningful only when a V.23 mode or 202 150 bps (or 75 bps) back channel mode is selected by MC0-MC4. This signal is not used in the 202 5 bps back channel mode.

3.4.9 Transmitted Data (TD)

Data bits to be transmitted are presented to this input serially; High (mark) corresponds to logic 1 and Low (space) corresponds to logic 0. This data determines which frequency appears at any instant at the TRANSMITTED CARRIER output pin (Table 3.3). No signal appears at the TRANSMITTED CARRIER output unless DTR is Low and RTS is Low.

3.4.10 Back Transmitted Data (BTD)

This line is equivalent to TRANSMITTED DATA for the main channel, except it belongs to the back channel. BTD is meaningful only when a 202 or V.23 mode is selected by MC0-MC4. For 202 5 bps back channel transmission of on/off keying, BTD should be fixed at a High level.

3.4.11 Carrier Detect (CD)

A Low on this output indicates that a valid carrier signal is present at the receiver and has been present for at least a time t_{CDON}. A High on this output signifies that no valid carrier is being received and has not been received for a time t_{CDOFF}. Carrier Detect looks for energy in the receive bandwidth. CD is Low when the receive signal is above a threshold limit V_{CDON} and High when the level of the received signal is below V_{CDOFF}.

For the Am79101 in the Call mode, CD=Low indicates a valid answer tone or call progress tone has been detected above the V_{CDCON} level. CD=High indicates there is no energy above the V_{CDOFF} limit.

3.4.12 Back Carrier Detect (BCD)

This line is equivalent to CARRIER DETECT for the main channel, except it belongs to the back channel. BCD is meaningful only when a 202 or V.23 mode is selected by MC0-MC4. For the V.23 back channel mode or the 202 150 bps (or 75 bps) back channel mode, BCD activates when either the mark or space frequency appears with sufficient level at the received carrier (RC) input.

For the 202 5 bps back channel mode, BCD turns on in response to a 387 Hz tone of sufficient level at the RC input. In this case BCD is equivalent to the Secondary Received Line Signal Detector for 202 S/T modems, or Supervisory Received Data for 202 C/D modems.

3.4.13 Received Data (RD)

Data bits demodulated from the RECEIVED CARRIER input are available serially at this output; High (mark) indicates logic 1 and Low (space) indicates logic 0.

Under the following conditions, this output is forced to logic 1, because the data may be invalid:

1. When $\overline{CD}$ is High
2. During the internal squelch delay at half-duplex line turn-around (202 and V.23 modes only)
3. During soft carrier turnoff at half-duplex line turn-around (202 and V.23 soft turn-off modes only)
4. When $\overline{DTR}$ is HIGH
5. When $\overline{RTS}$ is Low and $\overline{BRTS}$ is High in 202 and V.23 modes only
6. During the autoanswer sequence

Table 3.3a Frequency Parameters

Modem	Baud Rate (BPS)	Duplex	Transmit Frequency Space Hz	Transmit Frequency Mark Hz	Receive Frequency Space Hz	Receive Frequency Mark Hz	Answer Tone Freq Hz	Soft Turn Off Tone Hz
Bell 103 Orig	300	Full	1070	1270	2025	2225	–	–
Bell 103 Ans	300	Full	2025	2225	1070	1270	2225	–
CCITT V.21 Orig	300	Full	1180	980	1850	1650	–	–
CCITT V.21 Ans	300	Full	1850	1650	1180	980	2100	–
CCITT V.23 Mode 1	600	Half	1700	1300	1700	1300	2100	900***
CCITT V.23 Mode 2	1200	Half	2100	1300	2100	1300	2100	900***
CCITT V.23 Mode 2 Equalized	1200	Half	2100	1300	2100	1300	2100	900***
Bell 202	1200	Half	2200	1200	2200	1200	2025	900
Bell 202 Equalized	1200	Half	2200	1200	2200	1200	2025	900
CCITT V.23 Back	75/150	–	450	390	450	390	–	–
Bell 202 5bps Back	5	–	*	*	**	**	–	–
Bell 202 150bps Back	150	–	487	387	487	387	–	–

* ($\overline{BRTS}$ Low) and (BTD High): 387 Hz at TC
* ($\overline{BRTS}$ High) or (BTD Low): 0 volts at TC
* Meets CCITT R.20 frequency tolerance.

** 387 Hz at RC: $\overline{BCD}$ Low
** No 387 Hz at RC: $\overline{BCD}$ High
*** For V.23 soft turn off modes only

Frequency tolerance is less than ±0.4 Hz with 2.4576 MHz Crystal. Except Bell 202 which is +1.0 Hz (1200 Hz, mark).

Table 3.3b Timing Parameters

Am7910/101

Symbol	Description	Bell 103	CCITT V.21	Call Mode	Units
t_{RCON}	Request-to-Send to Clear-to-Send ON Delay	208.3	400	-	ms ±0.3%
t_{RCOFF}	Request-to-Send to Clear-to-Send OFF Delay	0.4	0.4	-	ms ±40%
t_{CDON}	Carrier Detect ON Delay	92-106	300-312	-	ms
t_{CDOFF}	Carrier Detect OFF Delay	21-40	21-40	-	ms
$t_{CDON'}$	Carrier Detect Call ON Delay	-	-	92-106	ms
$t_{CDOOFF'}$	Carrier Detect Call OFF Delay	-	-	21-40	ms
t_{RING}	Ring Delay	25	25	-	µs

Table 3.3b Timing Parameters (continued)

Am7910/101 (continued)

Symbol	Description	CCITT V.23	Bell 202	Bell 202 150 bps Back and CCITT V.23 Back	Bell 202 5 bps Back	Units
t_{RCON}	Request-to-Send to Clear-to-Send ON Delay	208.3	183.3	-	-	ms ±0.3%
t_{RCOFF}	Request-to-Send to Clear-to-Send OFF Delay	0.4	0.4	-	-	ms ±40%
t_{BRCON}	Back Channel Request-to-Send to Clear-to-Send ON Delay	-	-	82.3	-	ms ±0.64%
t_{BRCOFF}	Back Channel Request-to-Send OFF Delay	-	-	0.4	-	ms ±40%
t_{CDON}	Carrier Detect ON Delay	11.4-15.4	18-22	-	-	ms
t_{CDOFF}	Carrier Detect OFF Delay	5.4-13.3	12.4-23.4	-	-	ms
t_{BCDON}	Back Channel Carrier Detect ON Delay	-	-	17-25	17-25	ms
t_{BCDOFF}	Back Channel Carrier Detect OFF Delay	-	-	21-38	21-38	ms
t_{SQ}	Receiver Squelch Duration	156.3	156.3	-	-	ms ±3.3%
t_{STO}	Transmitter Soft-Off Duration	24*	24	-	-	ms ±2.3%
t_{RING}**	Ring Delay	25	25	-	-	µs

* Am79101 only
** Am7910 only

Am7911

	t_{RCON} ms ±0.81%	t_{RCOFF} ms ±13%	t_{CDON} ms	t_{CDOFF} ms	t_{AT} s ±0.44%	t_{SIL1} s ±0.42%	t_{SIL2} ms ±0.69%	t_{SQ} ms ±0.72%	t_{STO} ms ±0.81%	t_{RING} µs
Bell 103 Orig	25.0	0.52	9-20	4-23	1.9	2.0	–	–	–	–
Bell 103 Ans	25.0	0.52	9-20	4-23	1.9	2.0	–	–	–	25
CCITT V.21 Orig	25.0	0.52	10-20	10-20	3.0	2.0	75	–	–	–
CCITT V.21 Ans	25.0	0.52	10-20	10-20	3.0	2.0	75	–	–	25
CCITT V.23 Mode 1,2	8.0	0.52	3-7	3.4-11.3	3.0	2.0	75	9.0	8.0(b)	25
Bell 202	8.0	0.52	3-7	3.4-11.3	1.9	2.0	–	9.0	8.0	25

	t_{BRCON} ±0.64% ms	t_{BRCOFF} ±13% ms	t_{BCDON} ms	t_{BCDOFF} ms
CCITT V.23 back (a)	82.3	0.52	17-25	21-38
Bell 202 Back 5 baud	–	–	2-10	6-23
Bell 202 Back 150 baud	82.3	0.52	17-25	21-38

(a) Both 75 baud and 150 baud
(b) Soft turn-off tone is generated only for selected V.23 modes.

3.4.14 Back Received Data (BRD)

This line is equivalent to RD for the main channel, except that it applies only to the back channel. BRD is meaningful only for the V.23 or the 202 back channel modes. Under the following conditions this output is clamped High:

1. V.21/103 modes
2. $\overline{BCD}$ High
3. $\overline{DTR}$ High
4. $\overline{BRTS}$ Low and $\overline{RTS}$ High in V.23 or 202 150 bps modes only
5. During Autoanswer sequence.

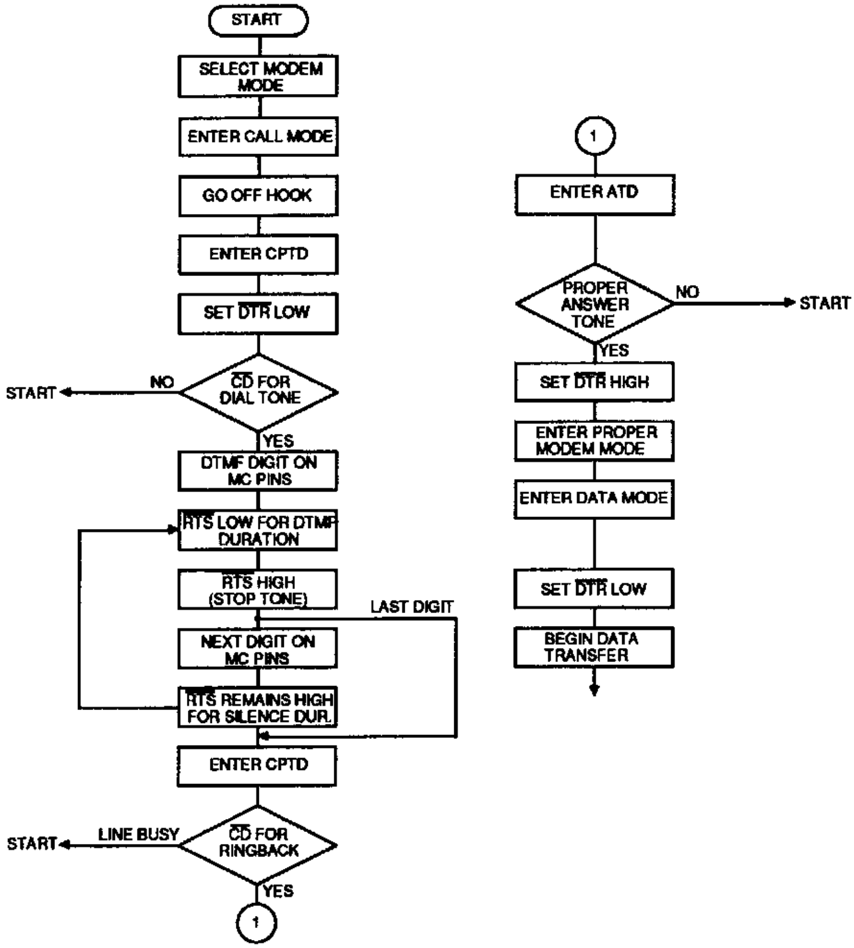


Figure 3.10a Autodial Flowchart

3.5 Am79101 CALL ESTABLISHMENT

The Am79101 supports Autodial and Autoanswer with the following features:

Call Progress Tone Detection (CPTD)
Dual Tone Multi-Frequency (DTMF) Tone Generation
Answer Tone Detection (ATD)
Answer Tone Generation (ATG)

First, the Mode Control (MC0-MC4) pins are used to set up the modem mode. Next, the Call mode is entered by setting DA/$\overline{CA}$=LOW (see pin description). $\overline{DTR}$ is then set LOW once at the beginning of the Autodial or Autoanswer sequence. $\overline{RTS}$ remains High except when tones are required to be transmitted. Now the MC pins can be used to select the above Autodial/Autoanswer features, as defined in Table 3.2b.

Autodial begins by using the CPTD function to detect the proper dial tone from the central office. The DTMF digit to be dialed is then input to the MC pins and the transmitter generates the corresponding DTMF tones at the TC pin. After the last digit is sent, CPTD is used again to detect the ringback signal. Following the detection of ringback, the ATD feature can be used to detect the answer tone from the called modem. Following detection of the correct answer tone, data transfer can now begin. Figures 3.10a and 3.10b show the Autodial flowchart and timings.

As a called modem, the Am79101 will be placed in the Call mode by the external microprocessor which receives a ring indication signal from the DAA. Autoanswer begins by using ATG. Figures 3.11a and 3.11b show the Autoanswer flowchart and timings.

3.5.1 Call Progress Tone Detection (CPTD)

The CPTD filter passes call progress tones from all the major Post Telephone and Telegraphs (PTTs). The cadences (temporal patterns) of $\overline{CD}$ will identify the following signals:

Primary Dial Tone
Secondary Dial Tone
Ringback Signal
Network Busy Signal
Busy Signal

Each country's PTT may have different requirements for the cadences of these call progress tones.

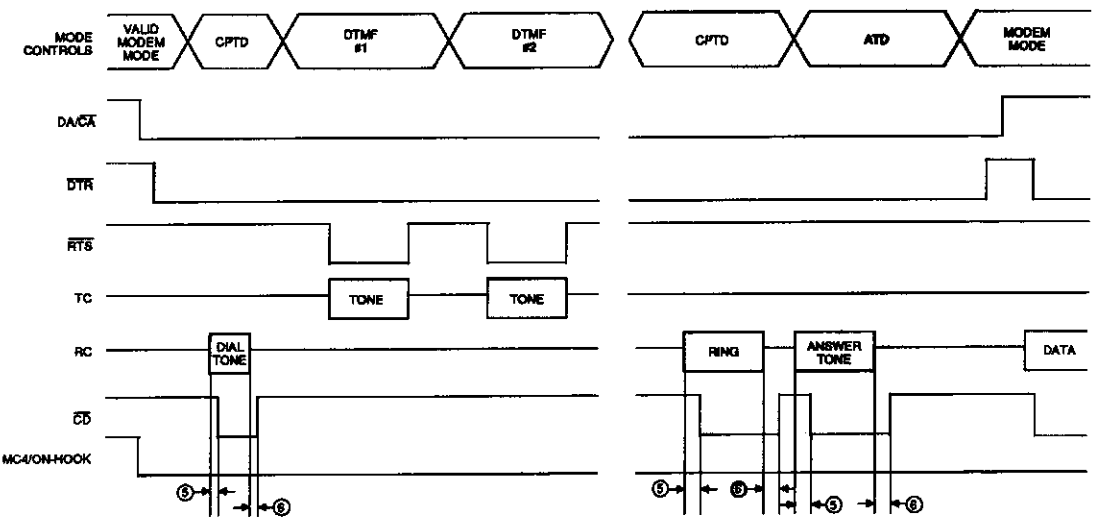


Figure 3.10b Autodial Timings

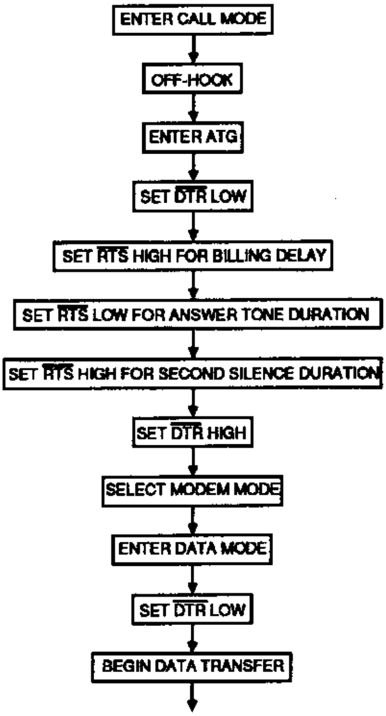


Figure 3.11a Autoanswer Flowchart

3.5.2 Dual Tone Multi-Frequency (DTMF) Tone Generation

The Am79101 transmitter can generate the basic 3 x 4 matrix, as shown in Table 3.4. According to the Bell specifications, the minimum duration for the DTMF digit is 50 ms. The minimum interdigit interval (silence) is 45 ms. The minimum time from the beginning of one DTMF digit to the next is 100 ms. It is common practice to extend the DTMF digit duration to insure detection by the central office.

Once in the Call mode, $\overline{\text{RTS}}$ controls the duration of the DTMF tone. With $\overline{\text{DTR}}$=Low and the desired digit applied to the MC pins, setting $\overline{\text{RTS}}$=Low will generate the DTMF tone at TC until $\overline{\text{RTS}}$ is set High. The user has complete control of the DTMF tone duration and the interdigit interval. Inputs to the MC pins should be changed while $\overline{\text{RTS}}$ is High and $\overline{\text{DTR}}$ is Low.

Table 3.4 DTMF Tone Combinations

		High-Group Frequencies (Hz)		
		1209	1336	1477
	697	1	2	3
Low-Group	770	4	5	6
Frequencies	852	7	8	9
(Hz)	941	*	0	#

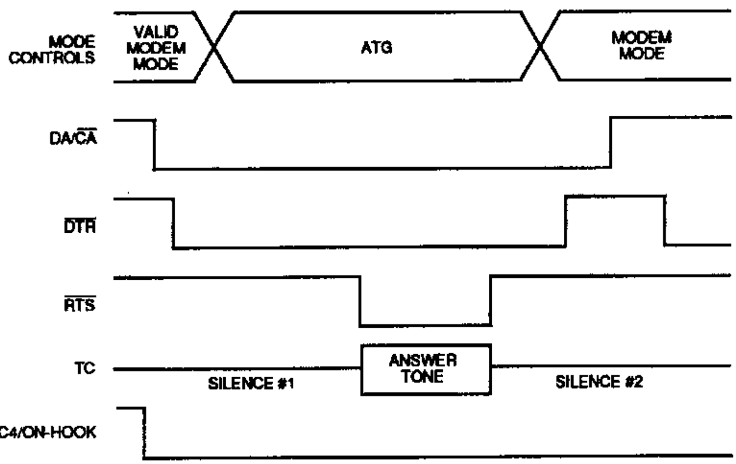


Figure 3.11b Autoanswer Timings

3.5.3 Answer Tone Detection (ATD)

The Am79101 receiver can detect answer tones for the Bell I03, 202, and CCITT V.21/V.23 modes. $\overline{CD}$=Low indicates the presence of a valid answer tone. RD distinguishes between the three answer tones as shown below. This feature allows mode matching with the remote modem. In the CCITT V.21/V.23 modes, X at RD indicates toggling between High and Low.

Modem Mode	Answer Tone Frequency (Hz)	$\overline{CD}$	RD
Bell 103	2225	Low	High
Bell 202	2025	Low	Low
CCITT V.21/V.23	2100	Low	X

The Carrier Detect Call thresholds (V_{CDCON}, V_{CDCOFF}) for valid answer tone detection are internally set to be higher than those in the Data mode. This eliminates the need for an external attenuator in the Call mode to prevent noise from being decoded as an answer tone.

3.5.4 Answer Tone Generation (ATG)

Answer tones are generated according to the modem mode selected, as listed in the ATD section. $\overline{RTS}$=Low controls the answer tone duration and $\overline{RTS}$=High controls the silence intervals. Silence interval before the answer tone is required by various governmental agencies for billing delay purposes.

3.6 AM7910/11 CALL SUPPORT

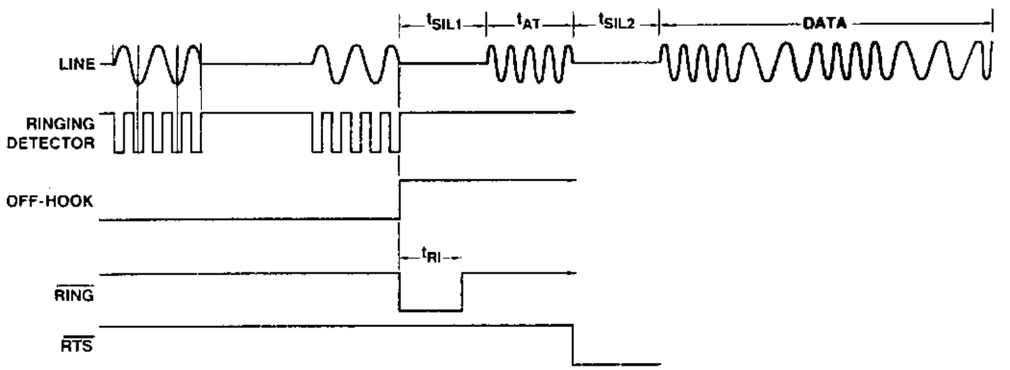
A Ring Detector Circuit can be used to provide Auto-answer capabilities for the Am7910/11. The Ring Detector circuit is explained in detail in Chapter 4. When a valid ring signal is detected, the output goes Low. This output can be tied to the $\overline{RING}$ pin on the Am7910/11.

After an incoming call is detected and the DAA is set in the off-hook mode, the $\overline{RING}$ pin to the Am7910/11 should be brought Low. The modem, set up in an answer mode, provides an answer tone after a silence interval. This silence interval is for billing delay. A sample timing diagram for this sequence is shown in Figure 3.12. It should be noted that once $\overline{RING}$ is brought Low, answer tone will never be generated again unless $\overline{DTR}$ is brought High and then Low again. $\overline{RTS}$ can be brought Low during the answer tone sequence so that data transmission can begin after the second silence period. The modem will ignore $\overline{RTS}$ until the answer tone sequence is complete. Optionally, the $\overline{RING}$ input can be tied High to disable the answer sequence.

3.6.1 Am7910/11 Answer Tone Detection

The Am7910 and Am7911 do not provide an ATD mode like the Am79101, but the receiver can be used to provide this function in the Bell 103 Originate mode.

After $\overline{DTR}$ is brought Low, the modem will be able to detect the three different tones generated by the remote Am7910/11 in this mode, because they all fall within the receive band. When an answer tone is detected by the modem, the $\overline{CD}$ pin will go Low. When a Bell 103 modem is providing the answer tone on the remote end, the RD pin of the modem will be High. If a Bell 202 modem is providing the answer tone, RD will be Low because of the 2025 Hz answer tone (a 'space' frequency for the Bell 103 originate modem). If a CCITT V.21 or V.23 modem is providing the answer tone, the state of RD will probably be Low, but the answer tone (2100 Hz) is very near the center of the receive band, so RD might not be stable. Regardless of whether RD is High, $\overline{CD}$ will always be Low. Note that $\overline{RTS}$ should not be Low while detecting answer tone, because it is not desired to transmit a carrier while in the Detection mode. Once answer tone has been detected, the ACU should put the modem into the



04811A-31

Figure 3.12 Auto-Answer Timing

proper operational mode (if not the Bell 103 originate) after taking $\overline{DTR}$ High. Data exchange may begin after detection of answer tone and the modem is put into the proper operational mode.

If answer tone is not detected by the modem after a user-determined length of time, it can be assumed that either the line or network is busy, the remote end did not answer, a wrong number was reached, or the network is having trouble. If this happens, the ACU may retry the call again later.

3.6.2 Am7910/11 Call Progress Tone Detection

The Am7910/11 does not have special modes for CPTD; however like ATD, the modem can be configured to provide assistance.

The V.23 back channel loopback mode may be used for detecting dial tone. The back channel filter is approximately 150 Hz wide and centered about 440 Hz, so the 440 Hz portion of the dial tone will fall nicely in the back channel receive band. Therefore, when the ACU generates an Off-Hook command to the DAA, it should also put the modem into the appropriate mode. The Back Channel Loopback mode must be used, because in a normal 1200 bps mode which uses a back channel, $\overline{RTS}$ must be Low in order for the back channel receiver to operate. If $\overline{RTS}$ is Low, a carrier will be generated by the modem. In the Back Channel Loopback modes it is not required that $\overline{RTS}$ be Low before the back channel receiver operates. When the Am7910/11 detects a precise dial tone on the line, the $\overline{BCD}$ pin from the Am7910/11 will go Low. It should be noted that the Back-channel Received Data pin (BRD) may be either High or Low when the receiver is detecting dial tone. BRD should be ignored while using $\overline{BCD}$ to detect dial tone.

If the central office does not provide a precise DTMF dial tone, it is still probable that the $\overline{BCD}$ pin on the modem will go Low in the presence of older dial tones. Most of the older dial tones are modulated signals which have frequency components within the range of the Am7910/11's back channel. However, the characteristics of the Progress Tones which are generated by older central offices are not completely predictable. (See the Bell system publication "Notes on the Network," Section 5, p. 114.)

Once the dial tone has been detected by the Am7910/11, the ACU may begin dialing. Before dialing begins, it is advisable to disconnect the Am7910/11 from the line; or at least turn off $\overline{DTR}$ to the modem. This will prevent dialing transients from reaching the modem. When changing modes on the Am7910/11 (in this case, from a Back Loopback mode to the Normal Operational mode), $\overline{DTR}$ must be High before the mode control pins are changed. Once the mode control pins are set properly, $\overline{DTR}$ may go Low. If the mode control pins are changed while $\overline{DTR}$ is Low, the resulting state of the modem is indeterminate.

3.7 DATA TERMINAL EQUIPMENT INTERFACE

The Am7910/11 can be interfaced to either parallel or serial I/O ports of standard data terminal equipment with a minimal external component count. Connection to an intelligent terminal or serial computer port is accomplished without extra computer interface components. A Universal Asynchronous Receiver/Transmitter (UART) is required to connect the Am7910/11 modem system to a parallel microcomputer bus. The UART handles the control/status of the modem to the microcomputer as well as the parallel-to-serial conversion of data sent over the bus.

3.7.1 Stand-Alone Modem

A stand-alone modem can be configured using the Am7910/11, RS-232C/V.24 line drivers and receivers, a phone line interface (DAA or acoustic coupler), and a 2-wire to 4-wire hybrid. A modem suitable for connection to a serial computer port is shown in Figure 3.13. Since the Am7910/11 interfaces only to TTL-level devices, RS-232C line drivers and receivers are required for connection to devices accepting standard RS-232C/V.24 voltage levels. The block labeled "DAA" in Figure 3.13 is a standard telephone line interface.

Automatic answering of the telephone can be assisted by the Am7910/11 in this arrangement. Once the DAA has been placed in the off-hook state by the user's interface, the DAA will assert $\overline{RING}$ on the Am7910/11. Answer tone will then be generated by the Am7910/11. This answer tone conforms to Bell and CCITT V.25 standards in duration, frequency and amplitude. Tying $\overline{RING}$ High on the Am7910/11 disables the generation of answer tone.

Full-duplex operation of this circuit requires that both receive and transmit baud rate clocks on the UART be the same. If half-duplex (CCITT V.23 and Bell 202) communications are desired with the low-bit-rate back channel, the receive and transmit baud rate clocks must be separate, e.g., in V.23 transmit at 1200 bps and receive at 75 bps or vice versa. No such restrictions apply if the back channel is not used.

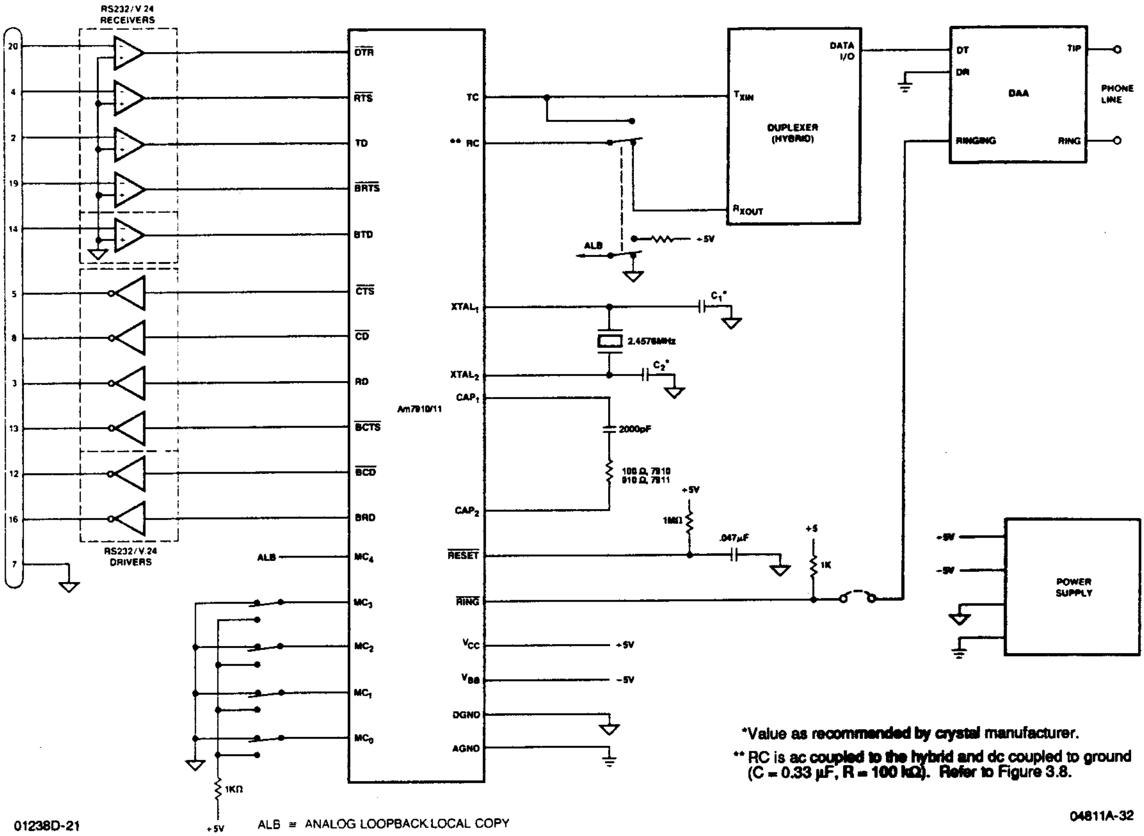


Figure 3.13 Stand-Alone Am7910/11 Application

3.7.2 UART Interface

The Am7910/11 can be interfaced to many standard eight-bit parallel data buses using the UART. The UART interfaces described in this document were designed for minimum parts count, low-cost, and a high degree of compatibility with existing microprocessors (the three major design criteria). Circuitry for an Am7910/11 modem used in conjunction with an 8251A UART (Figure 3-14) is described in this section. An Am7910/11 modem circuit using a MC6850 UART is described later.

Since the 8251A has separate receive and transmit baud rate clock inputs, it is possible to communicate in a Half-Duplex mode with a low-bit-rate back channel. Because the 8251A has only main channel RS-232C signals, simple gates are required as is shown in Figure 3.14 to select the proper pin connections to the Am7910/11's back channel signals ($\overline{\text{BRTS}}$, $\overline{\text{BCTS}}$, $\overline{\text{BCD}}$, BRD, and BTD) in this half-duplex operation. These gates are not required for full-duplex communication.

The 8251A is inhibited from read/write operations until $\overline{\text{CS}}$ is asserted. Once the 8251A is selected, a mode instruction written to the UART by the CPU places the 8251A into the Asynchronous Communications Mode, selects the baud rate clock multiplication factor, parity (odd, even, no), character length, and the number of stop bits attached to the data. All of this information is contained in one byte sent over the parallel data bus (D7-D0) when the Control/$\overline{\text{Data}}$ pin of the 8251A is High.

The Am7910/11 will only operate when $\overline{\text{DTR}}$ is Low. $\overline{\text{DTR}}$ of the UART can be controlled directly by writing a command word from the microprocessor. Once the Am7910/11 is operational, reception or transmission can begin. See the Am7910 or Am7911 Product Specification for the specific timing of each modem type.

The externally multiplexed Carrier Detect pins of the Am7910/11 can be connected to the $\overline{\text{DSR}}$ pin of the 8251A. The microprocessor can request the status of $\overline{\text{DSR}}$ and will thus be informed of the presence of a valid carrier.

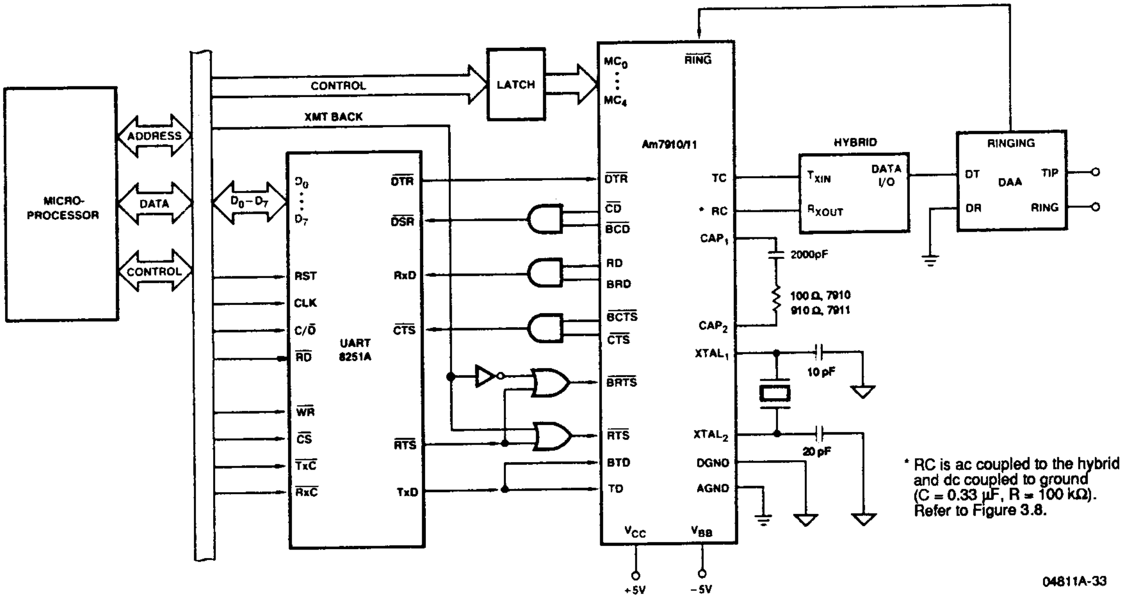


Figure 3.14 Am8251A UART Interface for Am7910/11

All the other control pins of the Am7910/11 are connected to their standard counterparts on the 8251A. Individual modem types can be selected by latching the data bus contents whenever a specific memory address is written. Note that the circuit in Figure 3.14 will work in either a Half-Duplex or a Full-Duplex Mode.

3.8 300 BPS FULL-DUPLEX SYSTEM CONFIGURATION

The system into which the FSK modems are designed can be configured in a variety of transmission modes and for many types of networks. This section describes how half-duplex and full-duplex circuits can be built for operation over both 2-wire PSTNs and 4-wire leased lines.

3.8.1 Am79101 – 300 bps, Full-Duplex Operation

Since a single baud rate (300 bps) can be used in either Bell103 or CCITT V.21 modes, it is possible with the FSK modems to make use of a single-chip microcomputer such as the 8051. The 8051 has a UART with a single baud rate generator on-chip.

The Am79101 is an ideal chip for a simple system implementation. The Am79101's RS-232C interface pins are all connected to the pins of the 8051 (see Figure 3.15) The eight set-up pins (DA/C̄Ā, MC0-MC4, D̄T̄R̄ and R̄T̄S̄) are connected to Port 1 of the 8051.

Off-hook is achieved by making MC4=0. MC4=1 will put the modem in Analog Loopback mode. In Analog Loopback, it is desired to be on-hook, so using MC4 as the off-hook control creates no problems. When the modem is not being used, MC4 should be set High (on-hook), MC=4.

The Am79101 can now be configured into the Call Mode to generate the DTMF tones required. If DTMF dialing is not permitted on a particular PSTN, the on-hook/off-hook relay controlled by MC4 can be rapidly made and broken to pulse-dial a number.

Automatic answering of incoming calls is also possible by monitoring the ring indicator circuit's output connected to the 8051's interrupt line. When ring occurs, the DAA can be placed off-hook and the Am79101 can enter the Call mode to generate the answer tone.

After the call is established or the call has been answered, the 8051 has complete control over all necessary signals. The modem can be set up for Bell 103 or V.21 and the 8051's UART will provide data transfer.

If local testing is desired, MC4 should remain High. TC and RC should be disconnected from the transformer so there is no voltage drop. Now the D̄T̄R̄ input can be set Low and data transfer in Analog Loopback mode can occur.

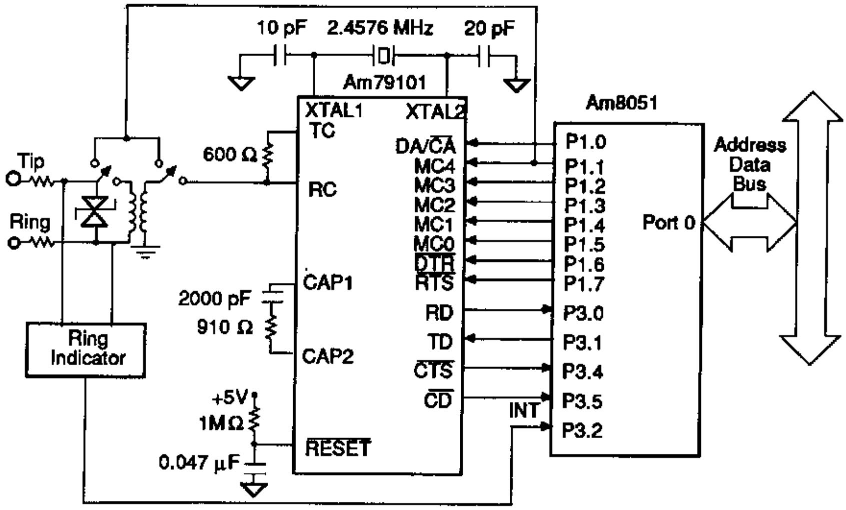


Figure 3.15 Am79101 300 bps Full-Duplex Circuit

3.8.2 Am7910/11 – 300 bps, Full-Duplex Operation

The Am7910/11 uses the same basic setup for 300 bps communication (See Figure 3.16). The connection to the phone line is controlled by a pin on the 8051 connected to the on-hook input of the DAA. Once an off-hook state is achieved, the remote modem can be automatically called by this system. A DTMF tone generator, such as the Mostek MK5089, can be used to dial the phone number. A pin on the 8051 controls the path of analog data to the DAA. This allows the selection of normal data transmission from the modem or DTMF from the MK5089. Pulse dialing can be made by rapid changes in the off-hook/on-hook relay.

Again, autoanswer is accomplished by monitoring the ring indicator input of the Am7910/11. When the DAA is placed off-hook, and the $\overline{\text{RING}}$ input of the Am7910/11 is asserted, an answer tone is generated. If local testing is required, the 8051 must set the MC4 input High, and TC must be externally connected to RC.

3.9 1200 BPS HALF-DUPLEX SYSTEM CONFIGURATION

The Bell 202 and CCITT V.23 Modes are 1200 bps half-duplex but they also contain a low-speed back channel. The low-speed back channels have two types: Bell 202 at 5 bps and Bell 202 or V.23 at 150 bps. The 5 bps back channel uses on-off keying and is used mainly for status, error information or request for line turn-around. The 150 bps back channel is used for FSK data.

3.9.1 Am7910/11/101 – 1200 bps, Half-Duplex Operation with No Back Channel

This set-up is a one-way communication channel. Data is passed to or from the modem at 1200 bps. Since there is only one baud rate and the handshake controllers are the main channel's $\overline{\text{RTS}}$, $\overline{\text{CTS}}$, TD, $\overline{\text{CD}}$ and RD, an application setup just like the 300 bps systems can be used.

3.9.2 Am7910/11 – 4-Wire, 1200 bps, Full-Duplex Operation

Dedicated network installations, known as leased-lines, are primarily used for data communications. A leased-line has the advantage that the line is a well-defined set of connections with repeatable characteristics. An attractive feature unique to 4-wire leased-lines is the ability to isolate the transmit signals from the receive signals, while using the same frequencies to communicate in both directions.

The FSK modems may be used in a 4-wire, full-duplex configuration by placing the modem in the proper Loopback mode. In Loopback mode, the transmitter and the receiver are internally configured to operate at the transmit frequencies for the modem type. The state machines operate the transmitter and the receiver independently in the Loopback mode.

Figure 3.17 shows the basic application for a 4-wire set-up using the Am7910/11. MC4 is tied high to enable the Loopback mode selected by MC0-MC3. In the Loopback

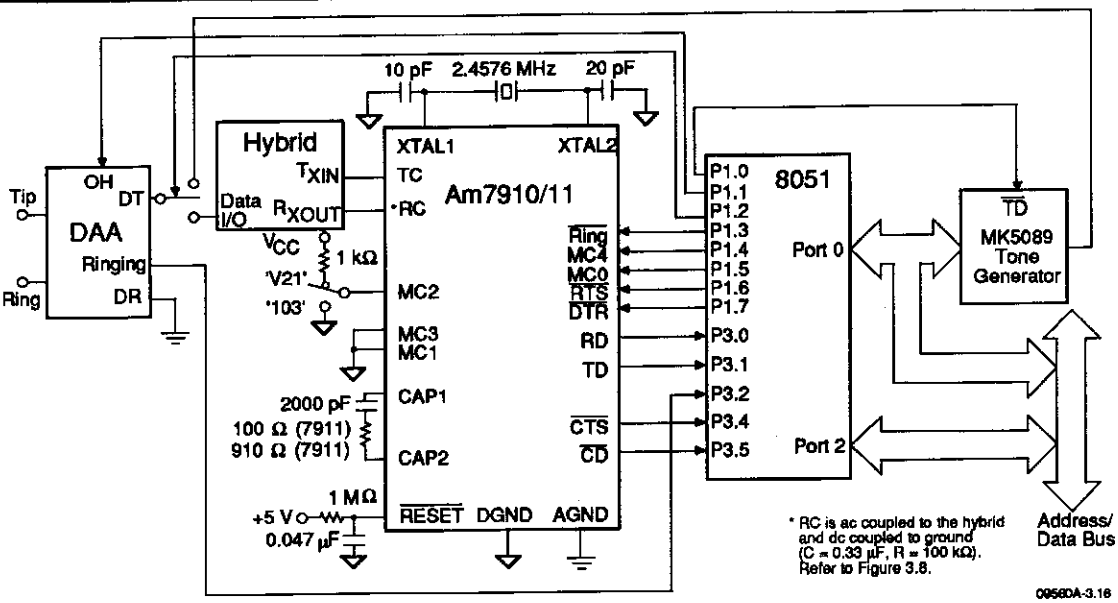


Figure 3.16 8051 Single Chip-Microcomputer Interfaced to Am7910/11

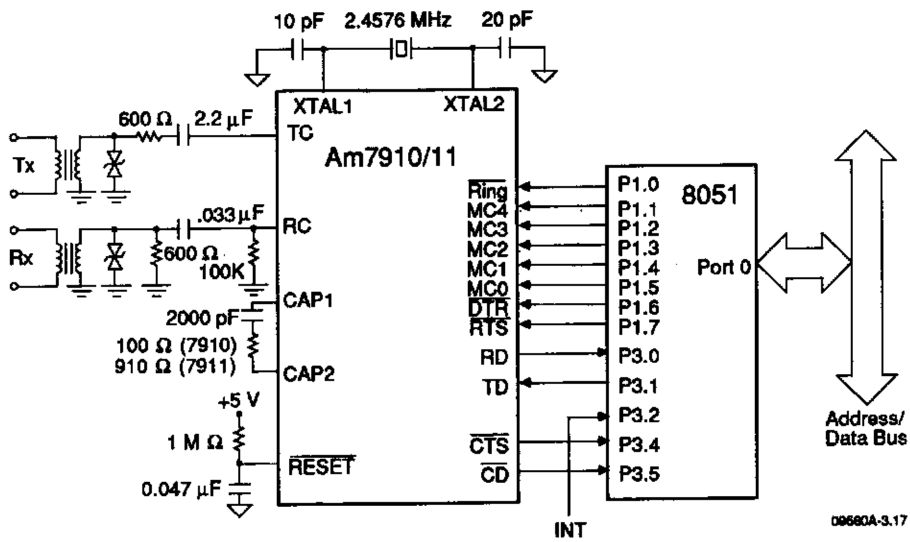


Figure 3.17 1200 bps 4-Wire Full-Duplex Circuit

mode, the state machines are independent allowing $\overline{CD}$ to be active when $\overline{RTS}$ is asserted in V.23 or Bell 202. Simultaneous transmission and reception at 1200 bps can occur in the Loopback mode. The handshake state machine's independence affects the squelch timing. When $\overline{RTS}$ is High, $\overline{CD}$ is not clamped for the normal T_{SQ} period. Squelch is not necessary in a 4-wire configuration, because there is no line turn-around procedure. Figure 3.18 gives a timing example for the 4-wire setup.

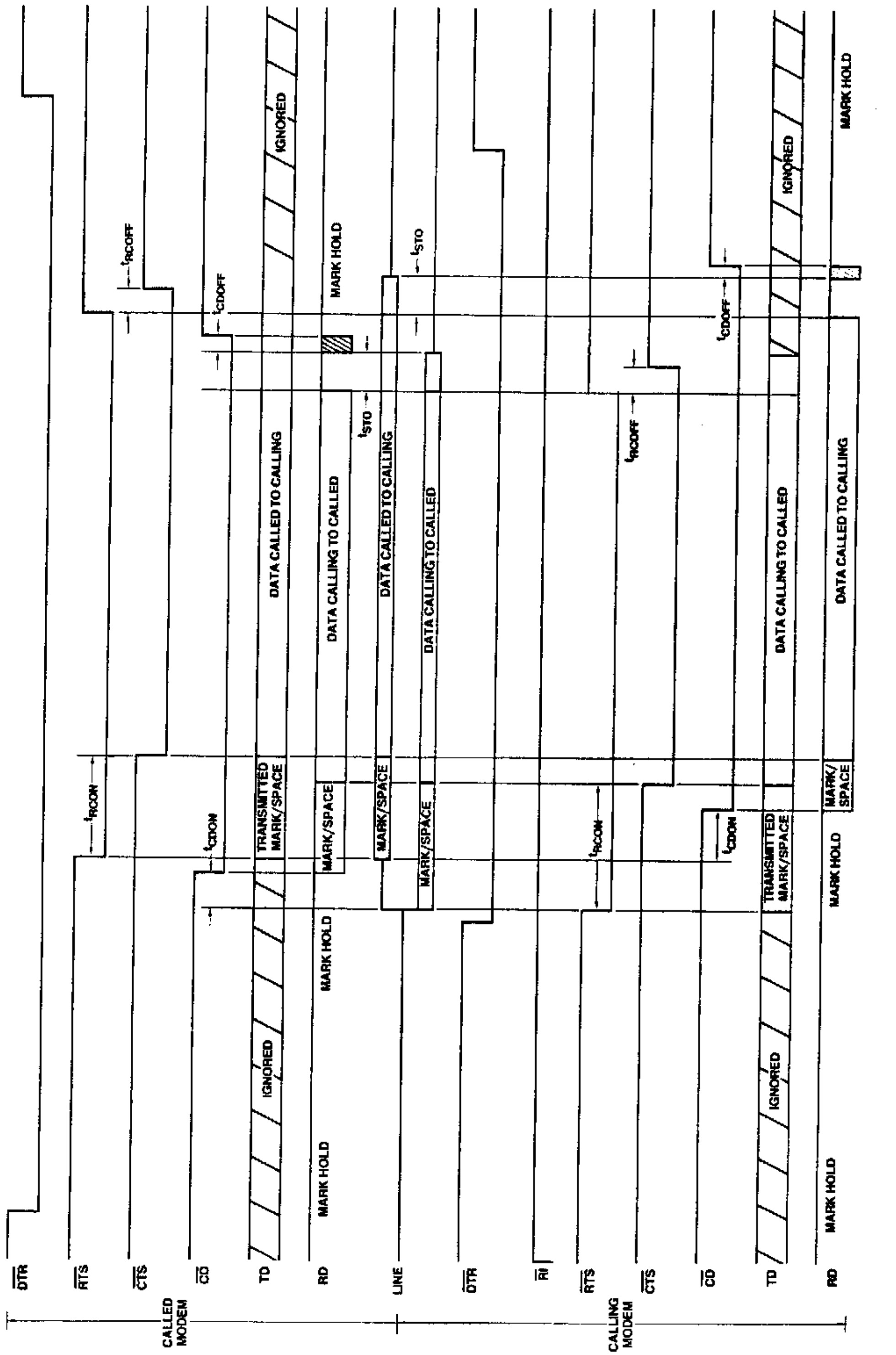


Figure 3.18 4-Wire, Full-Duplex Timing

3.9.3 Am7910/11/101 – 1200 bps, Half-Duplex Operation with 75/150 bps Back Channel

The Am79101 and Am7911's 150 bps back channel and the Am7910's 75 bps back channel are full data channels and operate much the same as the main transmission channel. Half-duplex communication with low-bit-rate back channel requires a more complicated hardware and software interface than the full-duplex transmission. The complexities occur for several reasons. Since the half-duplex modems are able to transmit at two different baud rates, two sets of RS-232C handshake pins (main channel and back channel) are available.

Most single-UART devices have only one set of RS-232C interface pins; therefore, the two sets of handshake pins must be multiplexed into these UARTs. One constraint on the single UART is the availability of only one baud-rate clock. Two alternatives exist: use an external serial communications controller or emulate a second UART with a separate port on the microcomputer.

An external serial communications controller, such as the Z8530, can be used with a single-chip microcomputer to provide a two-UART system having independent baud clocks for transmission and reception. Figure 3.19 shows a non-multiplexed data circuit utilizing the Z8530 as the communications interface to the Am7910/11.

The Am79101 uses the same application but has both the DA/$\overline{CA}$ pin and the simple line interface. The back channel handshake pins are connected to the corresponding B-Channel pins of the Z8530, while the main channel handshake pins connect to the A-Channel pins. The Z8530 can be used in a variety of different CPU systems such as the Z80, 8080, 8085, 6800, 6502, and 68000.

An alternative to the communications interface is to program an I/O port of a single-chip microcomputer to emulate the functions of a UART. This second UART will operate at the lower bit-rate for back channel communications.

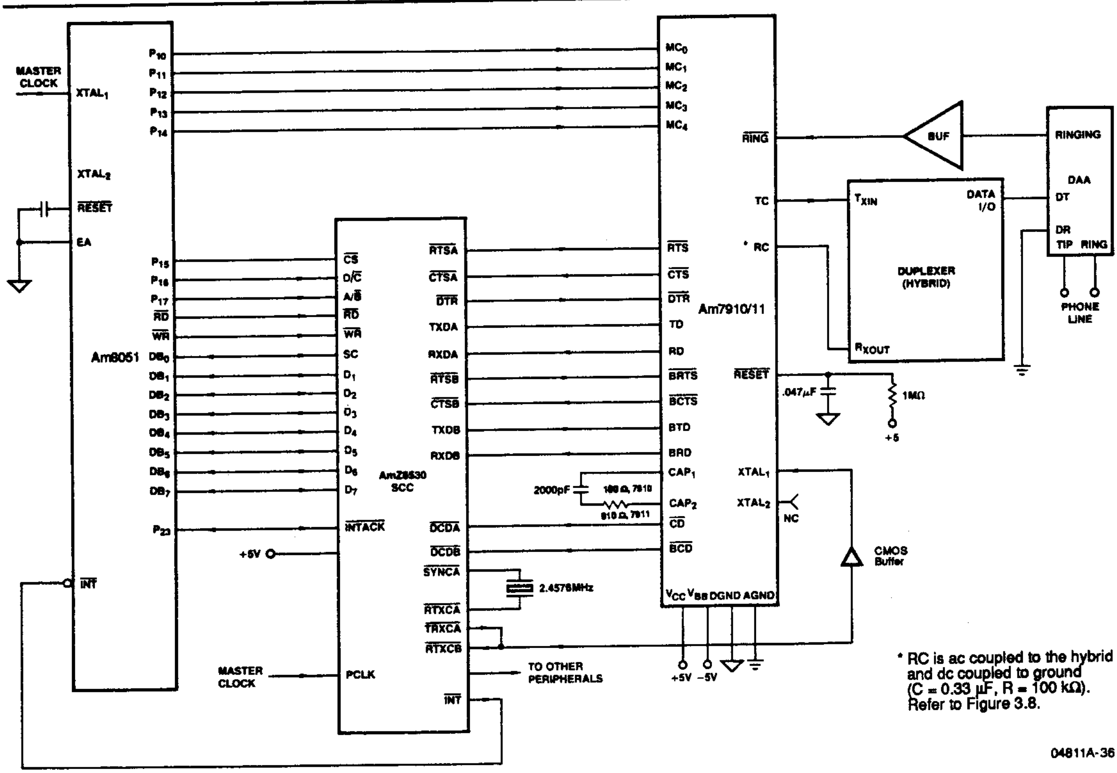


Figure 3.19 Two Independent Channels Provided by the Serial Communications Controller

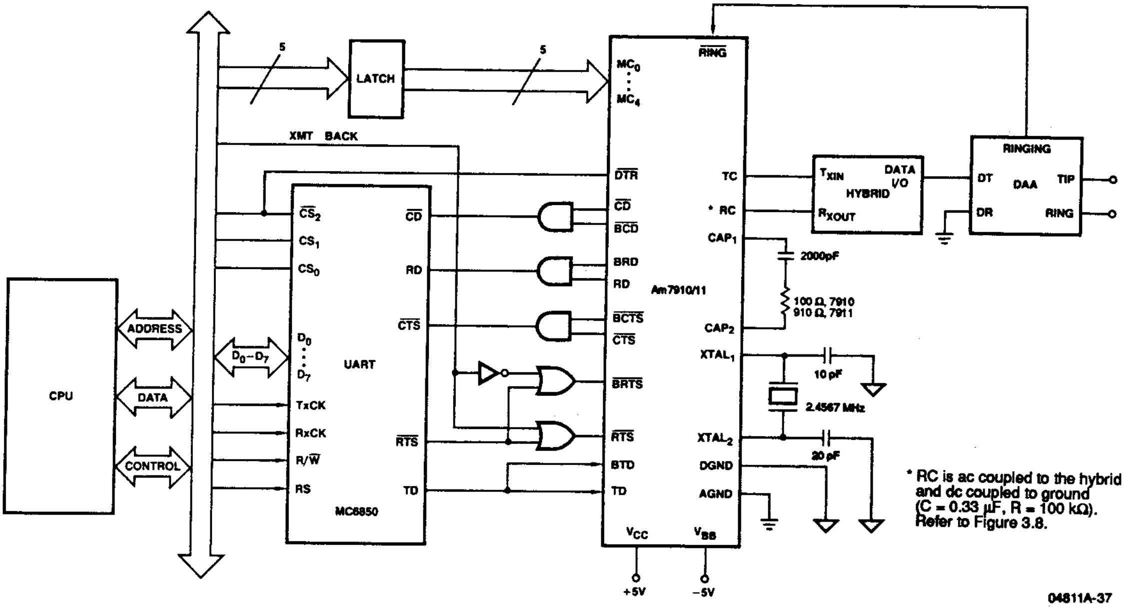


Figure 3.20 MC6850 UART Interface for Am7910/11

A cleaner solution is to use a UART that has separate transmit and receive clocks. Figure 3.20 employs a MC6850 UART as the interface between the Am7910/11 and the CPU. Note that this circuit is quite similar to the 8251A circuit shown previously.

The MC6850 has three chip select pins : CS0, CS1, $\overline{\text{CS2}}$, the former two pins are active High, while the latter is active Low. $\overline{\text{CS2}}$ can be connected to $\overline{\text{DTR}}$ so the entire UART/modem system will be activated on a Low input to this pin. As on the 8251A, the MC6850 has a single set of RS-232C input pins which may be connected to the multiplexed Am7910/11's main and back channel interface pins. Note also that the MC6850 has an input for carrier detect pins ($\overline{\text{CD}}$, $\overline{\text{BCD}}$). Two other features to note about the MC6850 are the separate transmit and receive baud rate clock pins (TxCK, RxCK) and the register select pin (RS) which is similar to the 8251A's CONTROL/$\overline{\text{DATA}}$ pin.

3.10 LINE TURN-AROUND SYSTEMS

Line turn-around is a communication protocol implemented at both the software and the hardware levels of a data communication system. A turn-around system takes advantage of the back channel available in the 1200 bps half-duplex modes. The back channel can give messages or interrupts to the system which uses the main channel. Based on this information, one can "turn around" the line so that the system at the other end can use the main channel. Control of the line turn-around is accomplished at the software level, while the actual transmission path reversal is done using the hardware interface.

AMD FSK modems have two types of back channels. The 150 bps back channel on the Am7911/101 and the 75 bps back-channel are full data channels at a slow speed. The Bell-202 5 bps back channel is used for interrupts (single messages) or some small number of messages (multi-messages).

3.10.1 75/150 bps Back Channel

Figure 3.21 is a diagram of an interface which has a straightforward line turn-around. The Carrier Detect pins ($\overline{\text{CD}}$ and $\overline{\text{BCD}}$) on the modem controls both $\overline{\text{DSR}}$ and $\overline{\text{CTS}}$ of the UART. This configuration causes the transmission of data to stop when loss of carrier is detected. $\overline{\text{BCD}}$ going High forces $\overline{\text{CTS}}$ and $\overline{\text{DSR}}$ High, stopping main channel transmission. $\overline{\text{CD}}$ going High forces $\overline{\text{BCTS}}$ and $\overline{\text{DSR}}$ High, stopping transmission on the back channel. Since it is not obvious at first glance that this is an advantageous configuration, an example protocol is presented:

The local transmitting modem has completed its transmission with an EOM character, and has taken $\overline{\text{RTS}}$ High. Carrier is removed and the remote modem $\overline{\text{CD}}$ goes High. The inactive $\overline{\text{CD}}$ at the remote modem forces both $\overline{\text{DSR}}$ and $\overline{\text{BCTS}}$ High at the 8251A. These actions

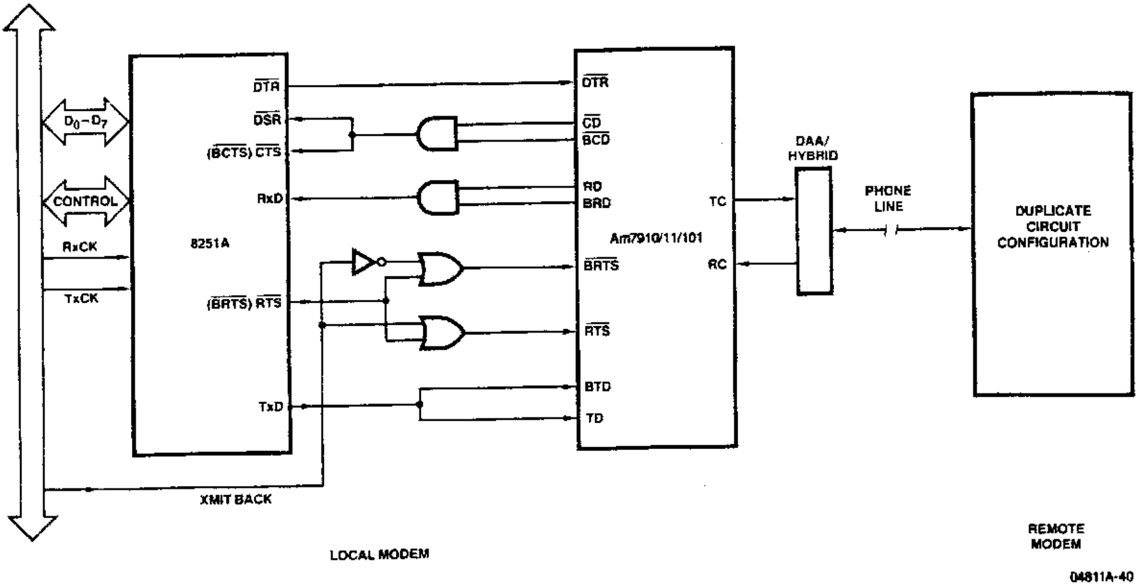


Figure 3.21 Half-Duplex Line Turnaround Block Diagram

lead the 8251A to remove BRTS and discontinue back channel transmission. At this point, no communications occur. Local BCD goes High forcing local CTS and DSR High. Now, the remote modem can begin the main channel transmission procedure by asserting RTS; a 'mark' frequency is transmitted. This carrier from the remote modem activates the local CD which forces BCTS and DSR Low. With DSR Low, the local modem will assert BRTS and generate a back channel carrier. The back channel carrier sets the remote BCD Low, which forces the remote CTS and DSR Low. With remote DSR asserted, corresponding TXD data now appears as a FSK carrier on the network, which completes the communications loop. The major drawback of this protocol is that it will not allow communication to occur in only one direction: both main and back channel carriers must appear on the network at all times.

3.10.2 5 bps Single Message System

The Bell 202 back channel can send a single message such as an interrupt or a request for line turn-around. The back channel operates differently from the main channel. BRTS becomes the back channel transmit data. When the local BRTS is Low, a signal is transmitted and the remote modem receives a Low on BCD. When local BRTS is High, no signal is transmitted and the remote modem's BCD is High. BCD acts as the received data pin.

In a system (see Figure 3.22), BRTS can be wired to a decoded CPU interrupt line, so that the CPU can interrupt the remote modem. On the receive side, BCD can be tied to a processor interrupt request line to receive an interrupt from the remote modem. Line turn-around can occur whenever an interrupt is received or transmitted.

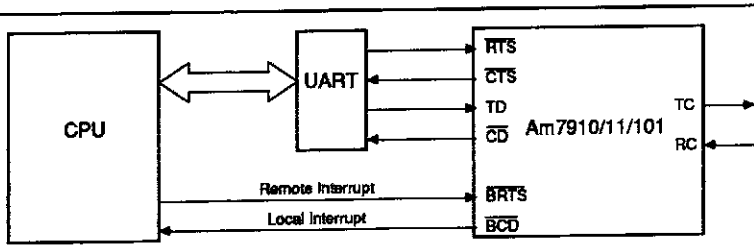


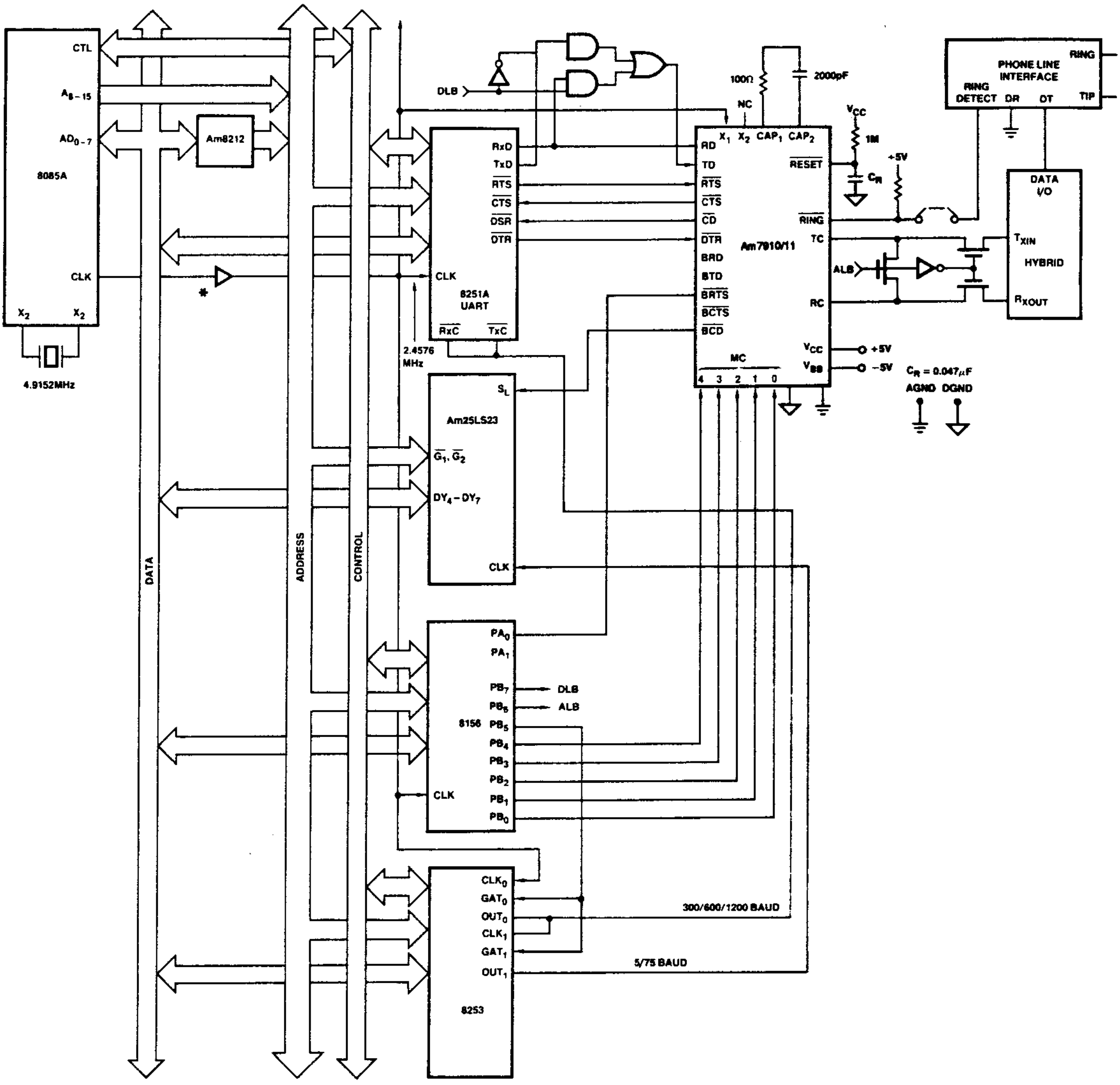
Figure 3.22 Interrupt Driven Bell 202 Configuration

When the local CPU sees an interrupt request ($\overline{BCD}$ LOW), it will bring $\overline{RTS}$ HIGH, turning off the remote modem's $\overline{CD}$. The remote modem can now assert $\overline{RTS}$ and begin transmission on the main channel.

3.10.3 5 bps Multiple Message System

If more than one type of message must be transmitted over the Bell 202 back channel, then a more sophisticated interface and protocol is required.

Since it is impractical in most cases to send or receive a byte of data at 5 bps (1.6 s/byte), most Bell 202 back channel messages consist of fewer bits. Byte-oriented UARTS such as the MC6850 and 8251A are generally not recommended for use with the Bell 202 back channel for this reason. An efficient substitute is a shift register. After a start bit has been sensed, the data bits can be shifted into the register followed by one or more stop bits. The shift register can be given an address in the



*Clock Buffer should be high speed CMOS to provide 3.8V HIGH level to Am7910 X_1 input.

04811A-39

Figure 3.23 Shift Register Bell 202 Configuration

processor's address space, thus allowing read and write capabilities of Bell 202 back channel data. Figure 3.23 shows a typical arrangement which allows 16 different four-bit messages to be recognized by the system.

Another protocol which works well with this Bell 202 configuration is initiated by an End Of Message (EOM) character. The modem transmitting on the main channel sends the EOM character to the remote system. Once the remote system recognizes the EOM, it awaits a loss of carrier at its Am7910/11/101 $\overline{CD}$ pin. The local modem sets $\overline{RTS}$ High and awaits transmission from the remote system. If no data is available for transmission at the remote system, $\overline{RTS}$ is never asserted. When data is to be sent from the remote system, remote $\overline{CD}$ must first go High. Then sensing clear line, the remote modem asserts $\overline{RTS}$ and soon commences transmission. The line turn-around process is complete for this protocol.

3.10.4 Line Turn-around Timing

Because the line turn-around is a timing overhead which must be incurred in a half-duplex modem, some effort should be placed in analyzing and minimizing the time spent in the line turn-around procedure.

When a line turn-around is being performed, the user-defined software of the remote system observes when $\overline{CD}$ goes High and then, after a delay (t_D), asserts $\overline{RTS}$ to commence transmission. The minimum time for t_D is dependent on the inherent delays of the modem type being used. The critical timing for Bell 202 and CCITT V.23 modems using the Am7910 is shown below.

Generally, as the local system sets $\overline{RTS}$ High at the end of the message, transients occur which may cause spurious spacing signals to be received at the remote system. For the Bell 202 case, the modem transmits a soft turn-off tone of 900 Hz for 24 ms after $\overline{RTS}$ goes High. The 900 Hz tone is closer to the 'mark' frequency (1200 Hz) than to the 'space' frequency (2200 Hz), so it is received as a 'mark' condition. Thus, the remote system receives a 'mark' hold for 24 ms (t_{STO}) after the main data reception is complete. When remote $\overline{CD}$ goes High, RD is clamped to a "mark."

Bell 202: $t_D = t_{SQ} - (t_{STO} + \min t_{CDOFF}) = 120$ ms

CCITT V.23: $t_D = t_{SQ} - \min t_{CDOFF} = 151$ ms

Thus, it would be advisable to keep t_D approximately 175 ms for safe operation.

3.11 Am7911's V.23/202 FAST RESPONSE TIMES

The Am7911 offers fast response times through a shortened t_{CDON} delay; it is possible that the t_{CDON} delay time is shorter than the processing delay for valid data. Because of this, it is possible to have glitches on RD up to 4 ms after the $\overline{CD}$ output has gone Low (see Figure 3.24).

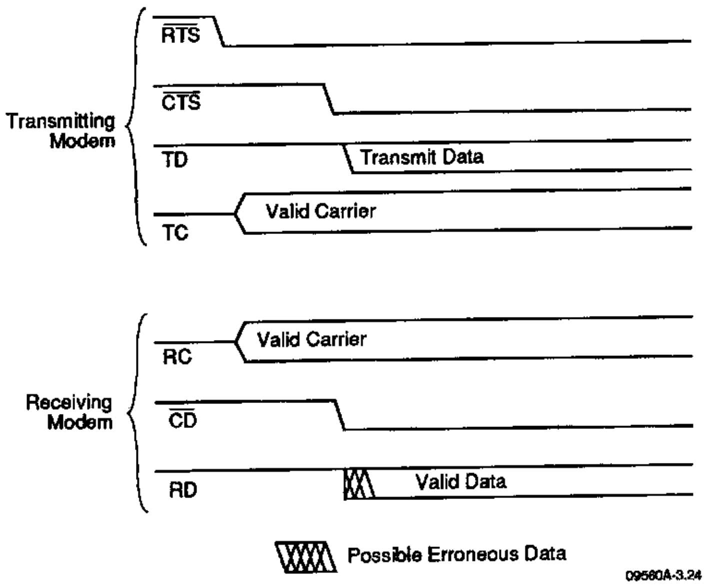


Figure 3.24 Am7911 V.23/Bell 202 Handshaking

One work around is in software: ignore RD until 4 ms after $\overline{CD}$ goes Low. If this is done, care must be taken that valid data from the transmitter is not missed. The transmitter can start valid data transmission after the t_{RCON} time.

3.12 CARRIER DETECT LEVELS

The $\overline{CD}$ pin indicates whether there is a valid carrier level within the receive band. The DSP algorithm looks at the amount of energy in a certain band and sets $\overline{CD}$ low, if the energy is higher than a specified value. The Am7910, Am7911 and Am79101 have different CDon and CDoff levels.

The incoming receive signal goes through the DAA and a hybrid during which there are changes in the carrier levels. The $\overline{CD}$ levels must be specified at specific points in this circuit. It is first assumed that the hybrid has been designed for the Am7910/11 so there is no loss from the telephone line level to the RC pin. The following chart gives the nominal levels at RC (or the line with no loss from line to pin).

To optimize the levels for CCITT specs, there must be an adjustment in the hybrid to provide gain so that, for example, a -43 dBm signal on the line looks like a -40 dBm signal at the RC pin of the Am7910/11. When the hybrid design is done, the final gain from the telephone line to the RC pin should be determined for the $\overline{CD}$ levels that the user is trying to meet.

Device		RC Level (dBm)	Hybrid Adjust (dB)	New Line Level (dBm)
Am7910	CDon	-40	+3	-43
	CDoff	-44	+3	-47
Am7911	CDon	-41.5	+1	-42.5
	CDoff	-47.5	+1	-48

In most applications, the Am79101 does not require an external hybrid. It's design assumes a -1.5 dB transformer coupling loss to achieve the following levels:

Device		RC pin (dBm)	Line Level (dBm)
Am79101	CDon	-45	-43.5
	CDoff	-49	-47.5
	CDCon	-37	-35.5
	CDCoff	-41	-39.5

3.13 RECEIVE DISTORTION MEASUREMENTS

The Receive Distortion test performed on AMD FSK modems uses a 63 bit pseudo-random binary sequence applied to the transmit data input. The output is verified at the receive data output. This is repeated for successive iterations while looking for the furthest transitions from the ideal edge. Resolution is one micro second. The distortion formula is shown in Figure 3.25.

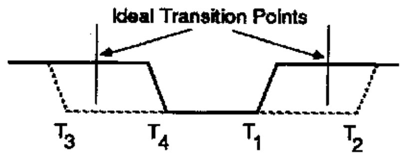
This distortion measurement provides a worst case indication of both bias and isochronous distortion, which are defined in Chapter 5.

3.14 FALSE TEMPERATURE PROBLEMS

The FSK Modems have very good performance over all temperature ranges. The modems come in Commercial (0°C to 70°C), Industrial (-40°C to +85°C)and Extended (-55°C to +125°C) temperature ranges.

If RC is not correctly ac coupled with a path for dc bias current, an apparent temperature sensitivity problem could be seen. As the modem heats up, the need for bias current to the Receive Carrier op-amp increases. It is necessary to ensure that the dc offset on the RC pin be kept within ±30 mV so that the internal op-amp is not starved for current. The usual indication of this current starvation is the RD pin locking up either High or Low.

The recommended approach to prevent this problem is to place a 100 kohm resistor from RC to AGND with a .033 µF capacitor between RC and the hybrid. If alternative values are chosen, care should be taken to ensure that the received signal is not filtered.



$$\text{Distortion} = \frac{(T_2 - T_1) + (T_4 - T_3)}{1/\text{Baud Rate}}$$

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Figure 3.25 Distortion Formula

3.15 COMMON PROBLEMS

The following is a summary of design considerations for the Am7910/11/101 and is intended to help designers avoid common pitfalls.

3.15.1 Power Supplies

As described in the Am7910, Am7911, or Am79101 Product Specifications, power supply dc tolerance is ±5%. The V_{CC} rise time must be a minimum of 5.0 msec to ensure proper oscillator startup, and power supply ripple must be less than 50 mV. It is important to note that excessive ripple will not only degrade Am7910/11/101's performance, but may also result in problems that are not obviously related to power supply disturbances.

3.15.2 Grounding

Analog ground (AGND) is a low-impedance ground for noise-sensitive analog circuitry, and should be as quiet as possible.

Digital ground (DGND) is a relatively noise-tolerant ground, and can be shared with other digital circuits on the same board.

No more than 50 mV offset should exist between AGND and DGND, and it is recommended that the two grounds be connected together at the power supply common to avoid ground loops. Good grounding and power supply practices are especially important if the Am7910/11/101 is in a noisy environment, e.g., near RF circuitry or a switching power supply. Proper use of decoupling capacitors and bus terminators can dramatically reduce ground noise.

3.15.3 Transmitted Carrier (TC) Pin

The maximum dc offset generated by the TC output of the Am7910/11 is ±100 mV; therefore, a blocking capacitor between TC and the line interface should be used.

3.15.4 Received Carrier (RC) Pin

A simple method of meeting the ±30 mV max dc offset specification of the Am7910/11 is to ac couple the Received Carrier (RC) input. A common pitfall is to isolate RC from the duplexor circuitry with a capacitor, without providing a path for dc bias current into RC (the RC input feeds an internal op amp). The recommended approach is to place a 100 kΩ resistor from RC to AGND, with a 0.033 µF capacitor between RC and the hybrid. If alternate values are chosen, care should be taken to ensure that the received signal is not filtered.

3.15.5 CAP1 to CAP2 Resistor Value

Am7910's recommended value is 100 Ω.

Am7911's recommended value is 910 Ω.

Am79101's recommended value is 910 Ω.

3.15.6 Analog Loopback

The Am7910/11/101's analog loopback modes condition the receive filters to match the transmit filters but do not provide an electrical path for the loopback. The user must connect TC and RC externally.

Analog loopback is also used to allow full-duplex Am7910/11 operation over 4-wire lines, with separate transmit and receive paths.

3.15.7 Reset

A Reset pulse is required after power up. Unexpected results can occur (e.g., incorrect transmit carrier level), if this pulse is omitted. Reset initializes Am7910/11/101's digital signal processing circuitry but does not directly affect the handshaking sequences.

In Figure 3.4 the parameter t_{DR} is specified to ensure the V_{CC} supply is stable before the minimum Reset pulse width t_{RL} is applied. The $\overline{\text{RESET}}$ input may start either Low or High, provided a minimum $\overline{\text{RESET}}$ Low time is observed after V_{CC} stabilizes. In the "Automatic Reset" case, an appropriate choice of time constant ensures that V_{CC} has stabilized (reached its 3.5 V level).

3.15.8 Data Terminal Ready ($\overline{\text{DTR}}$) – State Machine Initialization

When $\overline{\text{DTR}}$ is High, the Am7910/11/101's handshaking state machine is reset to its initial condition. This is the only way to reset the state machine and must be done after power up. The state machine does not automatically power up to a known state. If $\overline{\text{DTR}}$ is permanently enabled (Low), the state machine will simply run from wherever it powers up. This can result in abnormal behavior. For example: an unusually short $\overline{\text{RTS}}$-$\overline{\text{CTS}}$ delay can result from lack of $\overline{\text{DTR}}$ initialization.

3.15.9 Mode Changes

In order to change modes while the Am7910/11/101 is powered up, use the following sequence:

1. Take $\overline{\text{DTR}}$ High
2. Change mode inputs to desired configuration
3. Wait at least 100 µs
4. Take $\overline{\text{DTR}}$ Low

The mode inputs perform some hardware functions and are also sampled periodically by the state machine. If the mode inputs are changed without re-initialization through $\overline{\text{DTR}}$, unpredictable events may occur. For example, the Am7910/11/101 could operate halfway between two modes and thus be unpredictable.

The problem is similar to the case of a permanently Low $\overline{\text{DTR}}$; the Am7910/11/101 may work correctly or it may exhibit unexplained behavior. Both of these points are worth checking if the Am7910/11/101 is not functioning properly.

3.15.10 Request to Send ($\overline{\text{RTS}}$) and Clear to Send ($\overline{\text{CTS}}$)

It is not necessary to wait for $\overline{\text{CTS}}$ to go Low in order to transmit data, but if the message is transmitted and $\overline{\text{RTS}}$ goes High before $\overline{\text{CTS}}$ goes Low, soft turn-off tone and squelch delay will not occur. This is a consequence of attempting to run the Am7910 with leased-line timing parameters. If the message is sufficiently long , $\overline{\text{CTS}}$ goes Low and normal carrier operation will occur. The Am7911 is the recommended device for fast-handshake leased-line applications.

Another subtlety results if $\overline{\text{RTS}}$ goes Low in an attempt to transmit data immediately after the end of previous transmission (e.g., in a multi-point environment). The 156 ms squelch period t_{SQ} prevents transmission from occurring right away. This is not an Am7910/101 fault, but rather a consequence of pushing the handshake parameters in an attempt to maximize leased-line throughput. The squelch period can be reset by putting a minimum High pulse on $\overline{\text{DTR}}$ to shorten the inter-message delay. Again, the Am7911 is the recommended device for fast-handshake leased-line applications.

3.15.11 Crystal

The HC-33 crystal holder is recommended over the smaller HC-18.

When an external clock source is used via XTAL1, do not drive XTAL1 with a TTL driver and a pull-up resistor. The recommended external clock source is a CMOS driver (V_{CC} = 5.0 V).

CHAPTER 4

CHAPTER 4
DAA and Hybrid Design

Interfacing of any equipment to the Public Switched Telephone Network (PSTN) is governed by the national regulatory agencies. There are two different ways of interfacing modems to the phone line. One of these is to use a direct connection device called Data Access Arrangement (DAA). A DAA is a device which is physically connected to the phone line through an approved connector. In the U.S.A., the connector can take the form of an RJ-11 jack like that which is connected to most modular telephones.

The second interfacing scheme involves an acoustic coupler. An acoustic coupler uses the existing telephone handset and is not directly connected to the phone line.

Acoustic couplers suffer from a number of disadvantages relative to DAAs:

1. They require a standard telephone handset.
2. They are not easily adaptable to automatic calling and answering applications.
3. They will not work well in a noisy environment, because of the way the signal is coupled to the phone line.
4. They will work only with lower speed modems – typically they are not used above 1200 baud.
5. They are much larger than DAAs, because they must accommodate the telephone handset.

4.1 Am7910/11 HYBRIDS (DUPLEXOR)

The receiver and transmitter of the modem require a pair of wires each for physical connection. One pair of wires is available for standard PSTN connections. Therefore, a duplexor or electronic hybrid (also known as 2-wire to 4-wire converter) is used to attach the modem to the telephone network.

Imperfection in the isolation transformer makes impedance matching a complex task. Every transformer has insertion loss which reduces the input signal to the hybrid. Transformers for phone line isolation reflect incoming signals back into the telephone network at a reduced level. This is known as echo return loss. A set of standard isolation transformer requirements is given as follows:

Primary impedance: 600 Ω

Secondary impedance: 600 Ω (nominal)

Maximum DC current: 80 mA

Echo return loss: 20 dB

Frequency response: 300 - 3500 Hz ± 0.5 dB

Inversion Loss: 1 dB ± 0.5 dB (at 1.0 kHz)

DC resistance: Primary = 60 Ω ±10%
Secondary = 72 Ω ±10%

Harmonic distortion: 0.1% at 300 Hz, 80 mA

These specifications are not meant to be absolute requirements, but rather the normal limitations of an isolation transformer which would be used in a functional DAA.

Telephone networks have impedances which vary with frequency, length of the network connection, and type of cable used in the connection. Unfortunately, the routing of a call to a particular location varies from call to call. On the other hand, it is fairly easy to approximate the network terminating impedance over a range of distances.

Hybrid design presents several problems. First, the impedance of the telephone network is coupled through the DAA's isolation transformer. The hybrid should be designed to match the impedance of the network across the frequency band which is used for modulation to allow maximum power transfer.

The goals of an effective hybrid are:

1. A 6 dB loss from the Am7910/11's TC pin to the telephone network.
2. A 0 dB loss from the telephone network to the Am7910/11's RC pin.
3. A minimum signal transfer ratio from TC to RC.

While conditions (1) and (2) are only applicable to an off-hook situation where communications are occurring, item (3) must also be observed while the telephone is on-hook. To ensure that the receiver does not detect the coupled transmitter output as a valid carrier to be demodulated in the on-hook condition, there should be a loss from TC to RC. A 10 dB loss is in most cases sufficient to keep the transmit carrier from being detected and demodulated by the Am7910/11's receiver.

The circuit shown in Figure 4.1 matches the simple equivalent circuit used by the Bell network for local telephone network impedances (shown as Z_L). If the DAA uses a transformer which meets the criteria outlined above for an acceptable transformer, then, to simplify the explanation of our hybrid design, assume the transformer is ideal.

The balance condition which matches the telephone network load impedance is:

$$\frac{Z_B}{R_2} = \frac{Z_L}{R_1}$$

$R_1 = 600\Omega$
$R_2 = 22k\Omega$
$R_3 = 20k\Omega$
$R_4 = 2.5k\Omega$
$R_5 = 22k\Omega$
$R_6 = 22k\Omega$
$R_7 = 300\Omega$
$R_8 = 1550\Omega$
$R_9 = 39k\Omega$
$R_{10} = 11k\Omega$

$C_1 = 2.2\mu F$
$C_2 = 0.08\mu F$
$C_3 = 0.0022\mu F$

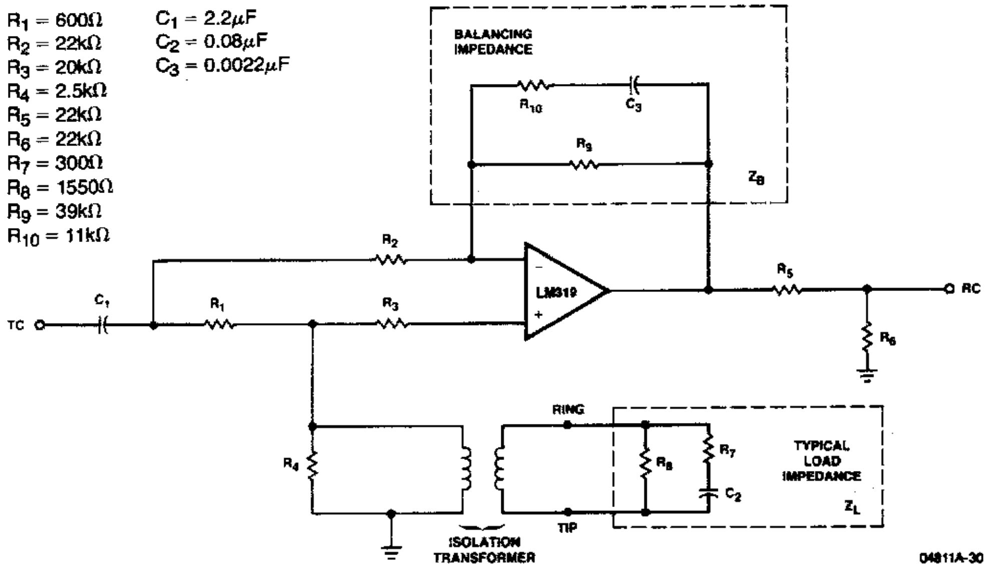

Figure 4.1 Hybrid Circuit

The balance network Z_B, is chosen to topologically match the load network Z_L, to simplify the computation of the components in Z_B.

To achieve the required 10 dB rejection from TC to RC in the on-hook condition, R_4 is placed in parallel with the secondary of the isolation transformer. This yields 10.2 dB loss in the on-hook case. The gain from RC to TC is given by:

$$\frac{V_{RC}}{V_{TC}} = \frac{R_2 Z_L - R_1 Z_B}{R_2(R_1 + Z_L)}$$

The current into both terminals of the op amp should be equal to avoid bias current at the Am7910/11's RC input. Resistor R_3 in Figure 4.1 places approximately equal bias currents at both op amp input terminals.

The resistors R_5 and R_6 act as an attenuator. The 6 dB loss from the op amp output to the RC pin of the Am7910/11 helps achieve two goals. To maintain a total of 10 dB rejection at the RC pin, 6 dB of attenuation of the TC output is necessary in the on-hook condition. The resistor attenuator also helps yield a unity gain from the RING connection at the network to the RC pin of the Am7910/11.

As discussed in section 3.12, the hybrid gain to RC can be adjusted to optimize the Carrier Detect levels. Since R_5 and R_6 act as an attenuator, their values may be adjusted to give the desired CD levels.

The op amp used in the hybrid should be a low-noise, high input impedance type with a large unity gain bandwidth. The op amp type chosen will greatly affect the overall system performance, so care should be taken in this component selection.

The previously described hybrid is an example of a typical Am7910/11 application. All hybrids must meet the DC offset requirements of the modem. Chapter 3 contains additional information regarding DC Offset and Carrier Detect Levels.

4.2 Am7910/11 NETWORK CONNECTION

Direct connection to the telephone network means that the modem is connected to the line through an approved interface device (DAA). There is a variety of requirements imposed on the DAAs to protect both the user and the network from harm. Figure 4.2 shows a typical DAA for the Am7910/11.

The single output of the hybrid is connected to the DAA which connects to the phone through TIP and RING.

The ring indicator circuit is used to activate the autoanswer function. The ring signal as presented to the network by U.S.A. central offices is a nominal two second "on", four seconds "off" signal at 120 V RMS and 20 Hz. The circuit provides a square wave output signal at the ringing frequency. The output can be connected to the RING input and when a low is detected, the modem will enter the autoanswer sequence.

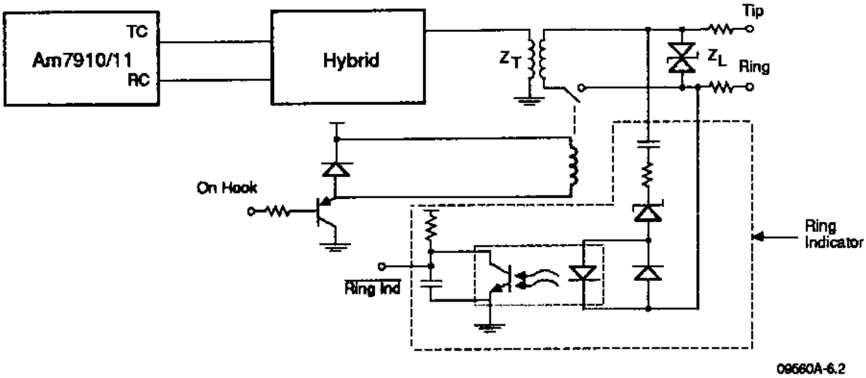

Figure 4.2 Am7910/11 Data Access Arrangement

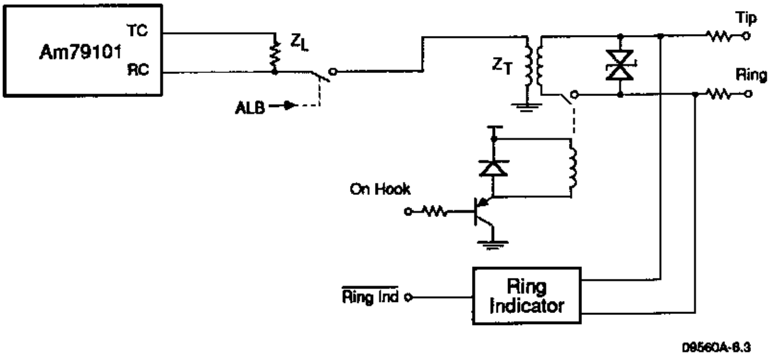

Figure 4.3 Am79101 Data Access Arrangement

For a detailed explanation of a DAA circuit refer to section 4.4. All information on the phone line side of the transformer can be directly applied to the Am7910/11.

4.3 Am79101 PHONE LINE INTERFACE

The Am79101 contains an on-chip hybrid which allows a simplified interface to the phone line. The hybrid subtracts the transmitted signal from the signal on the phone line so that the modem sees only the received signal. A typical phone line interface for the Am79101 is shown in Figure 4.3.

The ring indicator can be hooked to the system's processor. The signal going into the RC input contains some transmitted signal since it connects directly to the transformer. The transformer's Z_T and Z_L become a voltage divider for the TC signal. Z_L must be chosen to balance the transformer ($Z_L = Z_T$) where Z_T is the impedance of the phone line reflected back through the transformer. This balancing is required so that the transmitted signal is completely removed internally. Typically, a U.S.A. phone line can be modeled by a 600 Ω resistor. The balancing can be checked by observing the signal at the RC pin with no received signal from the phone line. This signal must be one-half of the amplitude and in phase with the TC output for perfect cancellation. Perfect cancellation will not be possible for all lines as Z_T will vary.

Analog loopback can be accomplished by connecting the TC output to the RC input or by disconnecting the transformer from RC so there is no voltage drop across Z_L (See Figure 4.3).

Four-wire applications and external hybrids are still possible. However, the internal hybrid will still be functioning. These applications require that there is one-half of the TC output in the signal at the RC input.

4.4 PROTECTIVE PHONE LINE INTERFACE DESIGN FOR VLSI MODEMS

4.4.1 Introduction

The circuit used to connect terminal equipment, including a VLSI modem, to the telephone line is known as the Protective Phone Line Interface (PPLI) or Data Access Arrangement (DAA). Functionally this interface provides coupling of the modulated voice-band signals to the telephone line as well as control points to emulate the telephone functions (on/off-hook conditioning, dialing and ring detection). More importantly, this interface is responsible for protecting the telephone network from harm resulting from the connection of terminal equipment.

In any given country, the national telephone authority provides a stringent specification to which equipment must be certified before it is connected to the telephone network. This application note describes the design of a low-cost, compact, protective line interface circuit tailored to applications utilizing VLSI modems and is intended to comply with the United States Federal Communications Commission's (FCC) Rules and Regulations, Part 68, and the Canadian Department of Communications' (DOC) Standard CS-03.

4.4.2 Alternative Approaches

Technically, the DAA and the PPLI provide two different design approaches to the interface problem. Traditionally, the DAA is considered to be a stand-alone interface which can be certified on its own as registered protective equipment. This allows a designer to connect terminal equipment, such as a modem, to the telephone line via a precertified DAA without requiring further certification. This approach allows a short time to market for prototypes and may be cost-effective for low-volume applications. However, due to its universal nature, the DAA must provide special functions, such as power level monitoring and signal squelching, to protect the telephone network against harm from improperly connected or malfunctioning terminal equipment.

A more cost-effective solution for modem applications is to include the modem VLSI with a custom line interface design and have the total solution certified as registered terminal equipment. This lessens the requirements of the PPLI compared to that of a universal DAA, because the PPLI does not need to protect the network against unknown terminals.

4.4.3 The Telephone System Environment

Before delving into the PPLI design details, a brief description of the telephone system environment is in order.

The telephone network consists of a hierarchy of switching nodes geographically spaced throughout the country. The hierarchy has a tree-like structure which defines the end-to-end routing path of any given telephone call. This structure allows a given switching node (office) to be optimized for a specific type of traffic, determined primarily by the geographic separation between endpoints. A switching office is classed according to its level in the hierarchy. For example, a subscriber's telephone is the lowest hierarchical level in a path and is usually connected to a Class 5 Central Office (CO) switch.

A telephone connection between two subscribers connected to the same CO would not require switching service from a higher level. However, a call from Canada to the U.S. could traverse a path from the originating Class 5 CO all the way up to a Class 1 Regional Center switch and then back down to the corresponding Class 5 CO at the answering end. The higher levels, Class 4, 3, 2, and 1, handle long-distance switching and are specialized to handle their respective interoffice signalling functions and traffic patterns. Class 5 central offices are specialized to handle the local subscriber loop which is the connection point-of-interest for a PPLI design.

4.4.4 The Subscriber Loop

Dial-up modems and telephone sets generally connect, over a pair of wires, to a Subscriber Line Interface of either a Class 5 CO or a PABX (Private Automatic Branch Exchange) in an office building. This subscriber loop consists of a pair of wires (Tip and Ring) which connect signalling information, voice-band audio and DC current from a CO or PABX to a subscriber.

To initiate a call the PPLI, under the control of the modem's microprocessor, provides a dc path for loop current to flow. This is the same effect as going off-hook by lifting the handset off of a telephone cradle. The flow of loop current is sensed by the CO which responds with a dial tone to indicate that dialing may commence. One of two methods is used to convey dialing information to the CO. Traditional pulse dialing is accomplished using a rotary dial which generates momentary interruptions in the dc loop current which, in turn, is sensed by the CO. Modem equipment uses touch-tone dialing which is provided at the subscriber end by a DTMF (Dual Tone Multi-frequency) generator.

DTMF dialing is the preferred method as it is more reliable, much faster, and more cost effective than pulse dialing. However, there is a large base of equipment still in operation today which recognizes only pulse dialing. Therefore, both methods are generally supported by modems with autodial capability. After dialing has been completed, call progress tones are returned by the CO to indicate the call status. Ringing is indicated if the called party is on-hook, busy if the called party is off-hook or trunk busy if the call could not be routed.

During an active operation and after a call is established, audio is transmitted over the subscriber lines superimposed on the dc loop current. This dc loop current serves two purposes. It indicates to the CO that the subscriber loop is active and it provides power to the subscriber's telephone set. To terminate a call, the subscriber goes on-hook and interrupts the loop current, which is sensed by the CO.

4.4.5 Electrical Requirements

The PPLI contains four circuits. They are the dc loop, AC transmission, transient protection and ringing detection circuits. Electrical requirements for each of these circuits are specified in the appropriate telephone authorities' specifications. For this design, the relevant documents are the FCC's Rules and Regulations, Part 68, and the Canadian Department of Communications' Standard for Terminal Equipment, Systems, Network Protection Devices and Connection Arrangements CS-03. Several good references are listed at the end of this paper.

The following is a summary of the relevant electrical requirements. They have been extracted from FCC Part 68, EIA RS-496 and DOC CS-03. The parameters fall within specification limits but do not reflect the worst case allowable limits for all conditions. In other words, some of the following parameters can be relaxed (at the spec corners, for example) without violating the certification requirements; however, this offers no apparent advantage to the selected design approach.

4.4.5.1 DC Requirements

The differential dc parameters of the modem are measured by applying a dc voltage across Tip and Ring and measuring the resultant current into Tip. The leakage currents to earth are measured by tying Tip and Ring together, applying the dc source from Tip and Ring to ground and measuring the resultant current into Tip and Ring.

On-Hook DC Resistance:

R > 10 MΩ for V < 100 VDC (Tip-Ring)
R > 30 kΩ for 100 < V < 200 VDC (Tip-Ring) (FCC)

R > 500 kΩ or I < 200 µA for V < 100 VDC (Tip-Ring)
I < 100 µA for V < 100 VDC (Tip-Gnd, Ring-Gnd) (DOC)

Off-Hook DC Resistance:
R < 200 Ω or does not increase by more than 25% of its initial value during the first 5 seconds after going off-hook. (FCC)

90 Ω < R < 300 Ω for 20 mA < I < 130 mA (DOC)

4.4.5.2 AC Requirements

The ac requirements are measured at Tip and Ring looking into the modem.

On-Hook Impedance (Ringer Impedance):

1600 Ω < Z < 40 kΩ for 15 < f < 70 Hz, 15 < V < 150 VRMS (FCC)

Z > 1600 Ω for f = 20 Hz and V = 45 VRMS
Z > 16 kΩ for f < 3.2 kHz and V = 1 VRMS
Z > 12 kΩ for f > 3.2 kHz and V = 1 VRMS (DOC)

Off Hook Impedance:

Source/terminating impedance 600 Ω ± 120 Ω (RS-496)

11 dB Return Loss (600 ± 400 Ω) (DOC)

Longitudinal Balance (measured from Tip to Ground and Ring to Ground, see Figure 4.6)

≥ 60 dB, 200 - 4000 Hz, On- or Off-Hook Power Levels (FCC)

Data: -9 dBm max into 600 Ω averaged over any 3-second interval. (FCC/DOC)

DTMF:
Low Group -10.5 dBm min
High Group -8.5 dBm min
Composite + 1.0 dBm max (RS-496)

4.4.5.3 Hazard Protection

Telephone terminal equipment must be capable of withstanding the following surge tests. If the equipment fails it must fail in a mode which does not cause degradation or harm to the telephone network.

Metallic Surge (one of each polarity) (Tip-Ring)
800 V peak 10 µs rise time, 560 µs fall time

Longitudinal Surge (one of each polarity) (Tip/Ring-Gnd)
1500 V peak 10 µs rise time, 560 µs fall time
Both On- and Off-Hook (FCC)

Metallic Surge (3 of each polarity)
1000 V peak 10 µs rise time, 1000 µs fall time
100 µs rise time, 1000 µs fall time (DOC)

4.4.6 Design Approach

The PPLI circuit is made up of four blocks (see Figure 4.4). These blocks are the dc loop current path, ac transmission, transient protection and ring detection.

4.4.6.1 DC Circuit (Figure 4.5)

4.4.6.1.1 On-Hook Resistance

The on-hook dc resistance is met by ensuring that the transient suppressing varistor (D8) turns on at no less than 200 VDC. The on-hook resistance is then determined by the reverse leakage currents of the varistor and the diode bridge. The sum of these must be less than 200 V/10 MΩ = 20 µA at 200 V.

4.4.6.1.2 Off-Hook Resistance

An active loop holding circuit sinks the dc current required to maintain the subscriber line in an off-hook condition while simultaneously providing a high ac impedance to avoid attenuating the signal being received by the modem. This approach blocks loop current from flowing through the transformer winding which minimizes the amount of transformer core material required to avoid saturation. The result is that a transformer may be utilized which is much smaller than one which is required to hold loop current.

The current sink is a Darlington transistor configured to look resistive for dc and exhibit a high ac impedance across Tip and Ring within the voice band. The Darlington is preceded by a bridge rectifier which provides polarity protection and proper operation in case Tip and Ring are reversed. Loop current ensures that two of the bridge rectifier diodes are forward biased causing only a small voltage drop for the ac signals. The circuit parameters are calculated as follows:

a) assume a min useable loop current of $I = 20$ mA

Need:
$2\ V < V_L < 6\ V$ to meet Off-Hook dc resistance (90 - 300 Ω)
$V_{CE} > 2\ V$ to keep Q2 out of saturation

$V_L = (I \times R4) + V_{CE} + (2 \times V_D)$

$R4 = (V_L - V_{CE} - 2 \times V_D) / I$
$R4\ min = (2 - 2 - 2 \times .06) / .02 < 0$
$R4\ max = (6 - 2 - 2 \times .06) / .02 = 140\ \Omega$

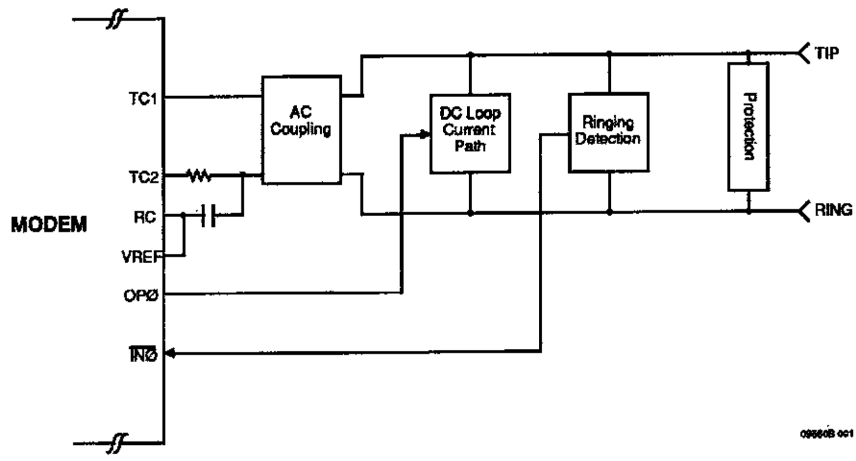

Figure 4.4 Modem Protective Phone Line Interface Block Diagram

Use the closest standard value less than R4 max. This minimizes the power that must be dissipated by Q1.

R4 = 120 Ω

The voltage divider ratio for the base is found by calculating the worst-case (lowest) base voltage required:

$V_B = (I \times R4) + (2 \times V_{BE}) = (0.02)(120) = 1.2 = 3.6\ V$

then determining the minimum supply voltage required to keep Q1 out of saturation.

$V_1 = (I \times R4) + V_{CE} = (0.02)(120) + 2 = 4.4\ V$

$V_B / V_1 \cong R3 / (R2 + R3)$
$3.6 / 4.4 = R3 / (R2 + R3)$
$R2 / R3 = 0.22$

Choose R2 and R3 to be large enough to contribute a negligible load across Tip and Ring but small enough to provide good bias stability, i.e., much smaller than the input resistance looking into the base of Q1, ≅ 120 x 10 kΩ.

R2 = 18 kΩ
R3 = 82 kΩ

Now the power dissipation of Q1 and R4 is determined using the extreme loop current case.

b) max loop current I = 130 mA

$V_B = I \times R4 = 2 \times V_{BE} = 0.13(120) + 2(1.0) = 17.6\ V$

$V_1 = V_B (R3 + R2) / R3 = 21.5\ V$

$V_L = V_1 = 2 \times V_D = 23\ V$

$P(R4) = I^2 \times R4 = 0.13^2 \times 120 = 2\ W$

V_L max = 23 V
R4 = 120 / 2 W

$V_{CE} = V_1 - (I \times R4) = 21.5 - 15.6 = 5.9\ V$
P(Q1) = 0.13 (5.9) = 0.77 W
P (Q1) = 1 W / 70 ° C

The D40C4 is a widely available (GE, Motorola, National), inexpensive Darlington transistor that meets the requirements outlined above.

D40C4

$BV_{CEO} = 40\ V$
$V_{CESAT} = 1.5\ V$
$P_{diss} > 1\ W$ @ 85° C
I_C max = 0.5 A

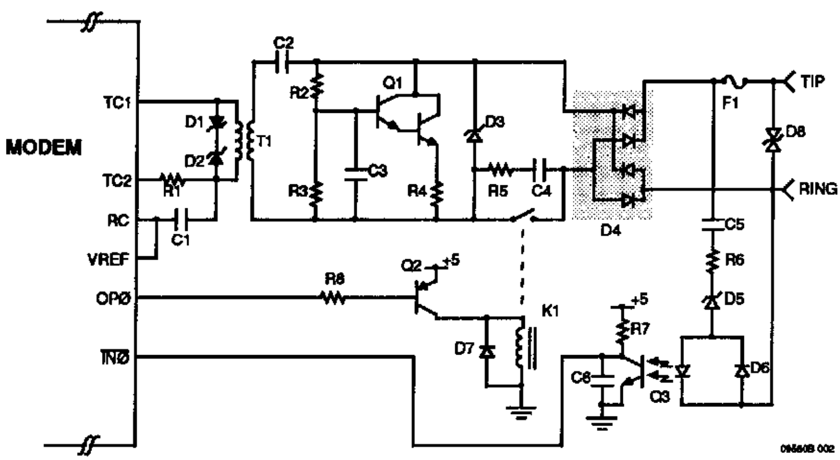

Figure 4.5 Modem Protective Phone Line Interface Schematic Diagram

c) Choosing C3:

C3 provides AC rejection for the bias circuit so that voice signals do not modulate the current sink. It must be large enough to provide adequate rejection above 50 Hz but small enough to provide a short time constant for dial pulsing. Choosing C3 = 1.0 µF gives good rejection above 50 Hz and a time constant of approximately 20 ms. Note that when the hookswitch relay is closed, there is an inrush of current through the transformer winding and blocking capacitor C2 which is in parallel with the RC time constant of the bias circuit. This will tend to speed up the rising edges of the dialing pulses.

Table 4.1. Parts List for Figure 4.5

C1	2.2 µF/10 V Tantalum Capacitor
C2	4.7 µF/30 V Tantalum Capacitor
C3	1.0 µF/30 V Tantalum Capacitor
C4	0.15 µF/250 V Capacitor
C5	1.0 µF/250 V Capacitor
C6	1.0 µF/10 V Tantalum Capacitor
D1, D2	1N4372 3.0 V Zener Diode
D3	P6KE27 Transzorb (General Semiconductor Industries)
D4	DF005 Bridge Rectifier
D5	1N978 51 V/0.5 W Zener Diode
D6, D7	1N4001 Rectifier Diode
D8	V150LA10A Metal-oxide Varistor (G.E.)
F1	0.5A Slow Blow Fuse
K1	LGK100-1-5 Relay (Standex)
Q1	D40C4 Darlington Transistor
Q2	2N3906 Transistor
Q3	4N36 Opto-isolator
R1	680 Ω 2% Resistor
R2	18K Ω 5% Resistor
R3	82K Ω 5% Resistor
R4	120 Ω/2 W 5% Resistor
R5	560 Ω/0.5 W 5% Resistor
R6	1K Ω 5% Resistor
R7	10K Ω 5% Resistor
R8	2.2K Ω 5% Resistor
T1	SPT-1104 Transformer (Prem Magnetics)

4.4.6.2 AC Circuit

4.4.6.2.1 On-Hook Impedance (Ring Detect Circuit):

The minimum ringer impedance, Z > 1.6 kΩ @ 70 Hz, is met with 1.0 µF in series with 1 kΩ (C5, R6). The 51 V zener diode (D5) serves two purposes. It helps prevent transients, due to dial pulsing for example, from triggering the ring detect circuit, and, in conjunction with D6, provides a high impedance over the voice band for low signal levels. This allows easy compliance with the on-hook impedance versus frequency requirement of DOC CS-03.

Ringing voltage is detected by the opto-isolator. Once per cycle of the applied ringing voltage a falling edge is applied to the interrupt driven input, INO, of the modem or microcontroller. Frequency detection can then be done in software to verify valid ringing. Diode D6 protects the opto-isolator diode against reverse breakdown. A dual (back-to-back) diode opto-isolator could be used, which would eliminate the need for D6. However, an interrupt would be generated twice per cycle which may be difficult for slower systems to service.

4.4.6.2.2 Off-Hook Impedance

Source and termination impedances are set by R1 (680 Ω) and the transformer impedances. This also sets the hybrid balance impedance in conjunction with TC1 and TC2 which are 180 degrees out of phase. This provides a balance point at the R1/C16 node which nulls the local transmit signal.

4.4.6.2.3 Power Levels

The modem should have its output voltages set to provide -9 dBm (max) power for data and -8.5/-6.5 dBm (typ) power for DTMF into the telephone line assuming a total of 4.5 dB of loss (3 dB matching, 1.5 dB insertion) between the differential output of TC1/TC2 and the telephone line. If a transformer with smaller loss is used, R1 can be increased to create more than a 3 dB matching loss to compensate. In this circuit the transformer loss is 1.0 dB, the active loop holding circuit and bridge rectifier contribute 0.3 dB (measured); therefore, the matching loss must be 3.2 dB, min.

Matching loss is $PdB = 10 \times \log (R / (R + R1))$ where R is the line impedance (600 Ω)

Therefore R1 min = 654 Ω, use R1 = 680 ± 2%.

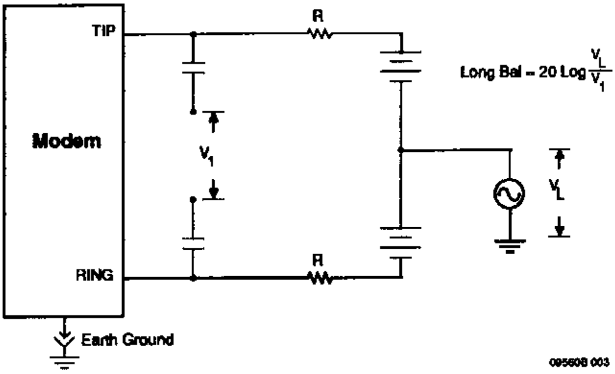

Figure 4.6 Longitudinal Balance Test

4.4.7 Longitudinal Balance

Longitudinal balance (common-mode rejection) is inherently met by keeping the entire line side of the circuit balanced with respect to ground. That is, there is no ground reference point on the line side of the transformer. Figure 4.6 illustrates how longitudinal balance is measured. The objective of the longitudinal balance specification is to minimize interference due to crosstalk and noise which is coupled to the telephone transmission lines in common-mode form.

The major role of the transformer is in providing longitudinal balance and dc isolation between the modem system and the telephone network. Its requirements are:

Turns ratio 1:1
Insertion Loss < 1.5 dB
Frequency Response 300 - 3500 Hz ± 0.5 dB
Return Loss > 11 dB
Longitudinal Balance > 60 dB
Total Harmonic Distortion < 0.5%

The Prem SPT-1104 was selected for this application.

4.4.8 Hazard Protection

Harm is any condition that degrades operation of the network due to failure of the terminal equipment or interface. Failure may be caused by a variety of abusive conditions such as large voltage transients induced by lightning and applied at the telephone leads or the ac main connection, or power line crosses to the telephone line. There are several protection devices in the PPLI to protect both the circuit and the network.

Transzorb D3 protects Q1 and is selected to clamp voltage spikes to a level less than the collector to emitter breakdown voltage of Q1. R5 and C4 form a "snubber" circuit which absorbs energy that is dissipated when the relay contact is opened. The energy generated by the collapsing dc magnetic field would otherwise be dissipated as a contact flash, decreasing the life of the relay and generating undesirable high voltage transients. Values for the snubber circuit are specified in EIA standard RS-496.

Metal-oxide varistor (MOV) D8 limits transients on Tip and Ring so that components with lower breakdown voltages may be used (K1, C4, D4, C5, D6). D1 and D2 protect the modem against spikes that swing outside of its voltage rails. Fuse F1, a slow-blow type, protects against power line shorts to Tip and Ring but does not blow during safe transients.

The pc board layout must maintain at least 1000 V isolation between the Tip and Ring conductors along traces from the telephone jack to D8. Additionally, 1500 V of isolation must be maintained between all conductors and system ground (or power) along traces, from the telephone jacks to the transformer.

4.4.9 Obtaining FCC and DOC Approval

The FCC and DOC regulations include other requirements such as mechanical stress testing, labelling and documentation. Both agencies provide a list of independent laboratories that have been approved to perform certification testing. Most of these labs also provide consulting services for approving terminal equipment under FCC Part 68, Part 15 and DOC CS-03 including circuit design consultation.

An approved laboratory should be selected and consulted early during the design phase since laboratories are generally the source of the most current specification issues and can provide tremendous help in critiquing a design. If everything goes well the first time, count on at least 90 days of lead time between submitting equipment for testing and receiving FCC certification. DOC approval may take longer.

The reference section at the end of this document lists relevant documentation available from the FCC and DOC as well as other useful references.

References

EIA Standard RS-496, "Interface Between Data Circuit-Terminating Equipment (DCE) and the Public Switched Telephone Network," May 1984.

Federal Communications Commission Rules and Regulations, "Part 68 - Connection of Terminal Equipment to the Telephone Network," December 1984.

Federal Communications Commission Rules and Regulations "Part 15 - Radio Frequency Devices," October 1982.

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Federal Communications Commission, "Instructions for Completing FCC Form 730, Application for Registration of Telephone Equipment."

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Government of Canada, Department of Communications, Terminal Attachment Program CS-03 Issue 6, "Standard for Terminal Equipment, Systems, Network Protection Devices and Connection Arrangements," January 25, 1986.

GTE, CHB-500 Issue 2, "Network Interface Manual for Registered Terminal Equipment," March 1982.

CHAPTER 5

CHAPTER 5
Performance Test Methods

5.0 MODEM PERFORMANCE EVALUATION

AMD's modems, unlike most integrated components, are a complete analog and digital system on a single chip. Thus, performance is necessarily measured at the system level. To simplify the evaluation task, individual measurements of the modem transmitter and receiver are desired. The modem transmitter generates a finite spectrum which is easily measured and compared to other transmitters. The merits of a modem transmitter are a narrow pass band with little or no amplitude ripple, nearly linear phase across the pass band, and a low stop band energy level. Harmonics and noise from the transmitter can be determined from overall system performance.

The modem receiver is not so easily measured because its performance depends mainly on the system environment. The test setup in Figure 5.1 is meant to closely emulate the typical data communications system and its surrounding environment.

5.1 TELEPHONE NETWORK CHARACTERISTICS

Telephone networks cause a considerable amount of distortion in the data transmission process. Almost every network is bandwidth limited to frequencies between 300 and 3400 Hz. This limitation is due to the Frequency Division Multiplexing (FDM) which occurs on many telephone networks. By modulating the baseband frequencies, 4 kHz slices of frequency are created throughout the frequency spectrum. Also, the physical nature of the network is such that amplitude is attenuated and phase is not completely linear across the 3 kHz bandwidth allotted to the users. Thus, frequencies between 300 and 3000 Hz are used for data transmission. Figures 5.2 and 5.3 show typical attenuation and group delay distortions for Bell networks and the CCITT V.56 recommendation for modem testing. Both types of distortion affect modem performance. The 300 bps modes of the Am7910/11/101 are fairly sturdy when placed on a distorted phone line and should not require any external

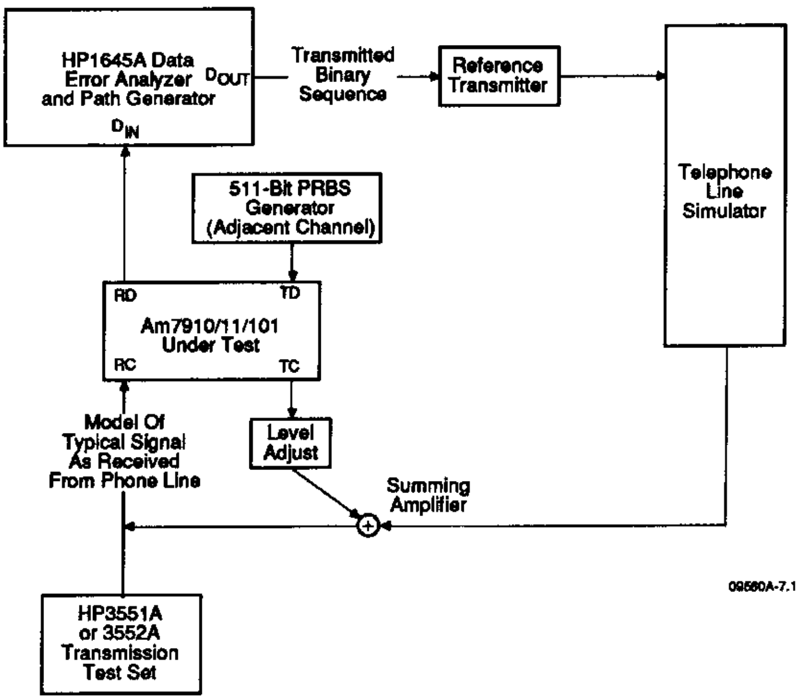


Figure 5.1 BER and Distortion Measurement Test Setup

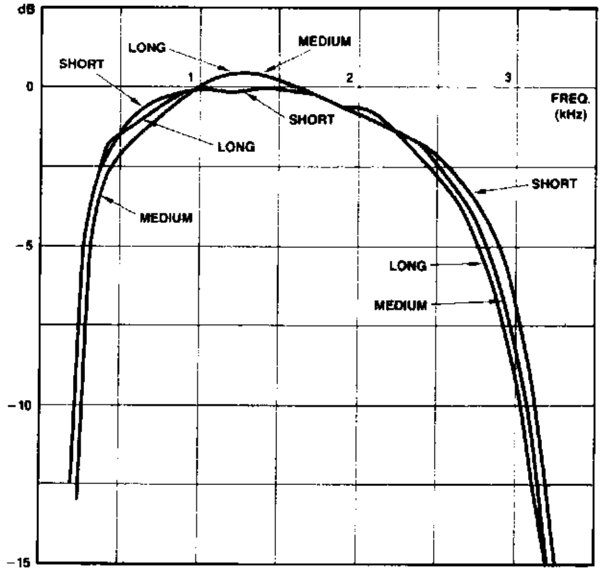


Figure 5.2 Bell Phone Characteristic Attenuation Distortion (relative to 1 kHz)

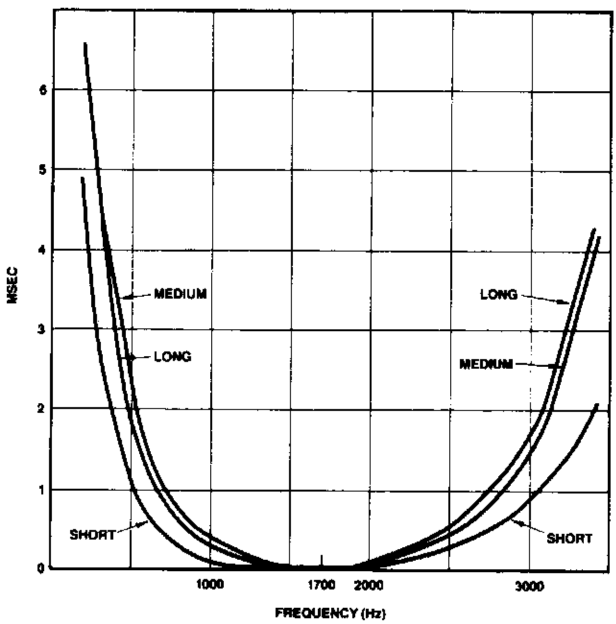


Figure 5.3 Bell Phone Characteristic Group Delay Distortion (relative to 1700 Hz)

compensation. The 1200 bps modes of the Am7910/11/101 are more susceptible to the network distortion than the 300 bps modes, but the modem can compensate for both amplitude and group delay distortion by selecting an equalized mode. The Bell 202 equalizer will compensate for a typical U.S. medium line, while the CCITT V.23 Mode 2 equalizer compensates for a typical European medium line.

Equalizers are filters with amplitude and phase characteristics designed so that when cascaded with the phone line response, the combined amplitude and group delay characteristics are flattened. Equalizers can be either fixed or adaptive. Fixed equalizers are filters with constant characteristics. A single amplitude/group delay characteristic is designed to equalize a variety of lines. Typically, fixed equalizers are used in modems operating at 0 to 1200 bps. Adaptive equalizers are time varying filters. Their characteristics vary over time to change the amplitude and group delay of the filter to compensate for the slowly varying amplitude and group delay of the phone line. Typically, adaptive equalizers are required in modems operating at 2400 to 9600 bps. The Am7910/11/101 uses digital filters to implement its fixed equalizers.

5.2 MODEM TEST SETUP

A typical bench test setup for the Am7910/11/101 is shown in Figure 5.1. A Binary Pattern Generator and Receiver/Analyzer for the returning data is necessary to measure modem performance. The HP1645A Data Error Analyzer generates a Pseudo-Random Binary Sequence (PRBS) at DOUT referenced to a synchronous clock. At some small increment of time later, the HP1645A expects the data to appear at DIN. This method of synchronously transmitting through an asynchronous modem is known as isochronous transmission. The PRBS generated by the HP1645A is modulated by a Reference Transmitter into an analog signal. Since no modulator is a perfect transmitter, some distortion to the analog waveform occurs at this point. However, due to the presence of equalization filtering in the transmitter and the difficulty of deriving precisely defined, repeatable measurements from the analog transmit signal, most performance efforts focus on measuring combined transmitter/receiver performance in the presence of various line types and impairments.

Several devices are available to simulate the characteristics of the telephone network. Among the ones used to test the Am7910/11/101 are the Wandel and Golterman TLN-1, DLZ-4 and the A.E.A. S3. These telephone network simulators shape the transmitted waveform so attenuation and phase (or group delay) characteristics of the transmitted signals liken to those from an assortment of networks.

Noise is present on the network and is generated by the reference transmitter in the communications system. In the test setup, the noise source in the Bradley 2A, or an externally filtered noise source, serves as a replacement for the network noise. The noise filter used to measure the Am7910/11/101's performance is band limited to frequencies between 300 and 3400Hz.

In most systems, the local transmitter is present at the receiver input in an attenuated form. By placing simulated data (normally a 511-bit PRBS) at the transmitter input, an analog carrier can be generated. By adjusting the carrier level and adding this waveform to the simulated telephone network carrier, the receiver under test will have a completely modeled signal of a standard modem receiver input.

5.3 ASYNCHRONOUS MODEM DISTORTION

The transitions in the data received by the data terminal equipment (CPU, computer, terminal) will not be exactly at the points where they occur in the test transmission, because modulation, network imperfections and demodulation affect the analog carrier adversely. Additionally, transitions in both the modulation and demodulation processes are not instantaneous, producing another form of distortion in the digital waveform. While the distortion is manifested only in asynchronous transmission, there are various names and meanings given to this digital distortion. In fact, the same name may imply a different test definition or test condition from one vendor to another. For this reason, the modem user must take great care when comparing test results to ensure that both the mathematical test definition and the test conditions are the same for both modems. Two of the best known forms of digital distortion are bias distortion and isochronous distortion.

5.3.1 Bias Distortion

Bias distortion is a measure of whether a modem has a preference for demodulating MARK or SPACE, when receiving an alternating MARK-SPACE pattern. The mathematical definition is:

$$\% \text{ Bias Distortion} = \frac{T_M - T_S}{T_M + T_S} \times 100\%$$

as shown in Figure 5.4a.

Ideally, the MARK and SPACE periods should be equal, but the duty cycle of the receiver output will indicate if the modem has a preference for demodulating one over the other. This test is useful both as a means of comparing modems, as well as a method for adjusting older modems with discrete components. The bias distortion test has the advantage of being relatively simple, but it does not reveal to what degree the modem will displace the data edges when receiving a pseudo-random pattern, which corresponds more closely to actual operating conditions.

5.3.2 Isochronous Distortion

Isochronous Distortion, as defined by specification RS-334-A, is a measure of the absolute latest transition in a data stream and the absolute earliest transition in a data stream when compared to the ideal sample instants. This type of distortion is data content dependent; it should be measured using a pseudo-random data pattern. The mathematical definition is:

$$\% \text{ Isochronous Distortion} = \frac{T_L - T_E}{T_{bit}} \times 100\%$$

as shown in Figure 5.4b.

If a transition occurs before the ideal sampling instant, then the sign of the period (T_E) should be negative. Note that this measurement definition is based upon the worst case transitions, and does not reveal the average departure of the received data edges from the ideal instants. For this reason, it is common to encounter other test definitions which attempt to provide an average distortion measurement.

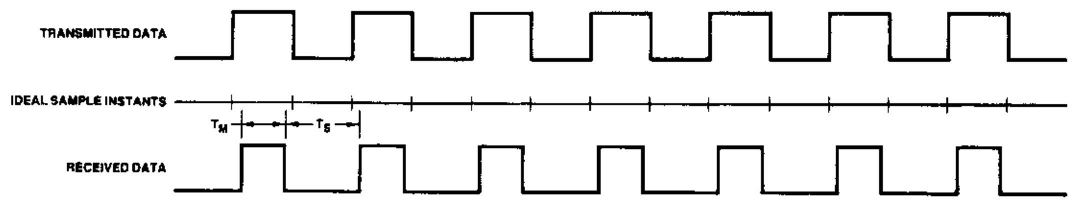
The distortion formula given in section 3.13 provides a worst case indication of both bias and isochronous distortion. The point to keep in mind is that there is no single standard way to measure asynchronous modem distortion; modem users must ensure that they are not comparing results from dissimilar tests, even if the tests have the same name.

5.4 BIT ERROR RATE ANALYSIS

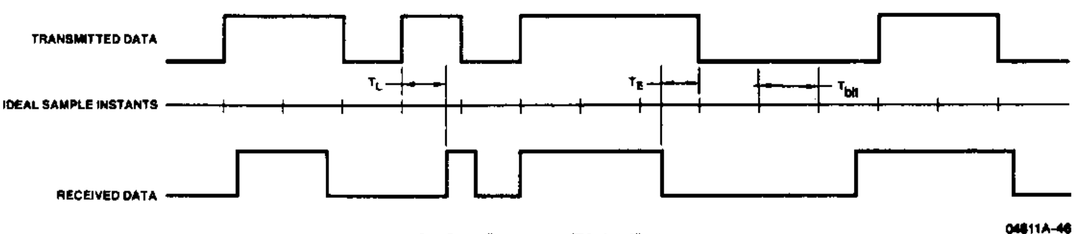
Bit Error Rate (BER) analysis is the most popular means of comparing different modems. By sending a large number of bits through the simulated communications system, the performance of the modem can be measured at the receiving end by the data error analyzer. In this manner, the bit error rate or Pe (Probability of Error) for a modem can be calculated for the test conditions used. A PRBS is generally transmitted so that the modem is tested using all of the data patterns that can be expected under normal operating conditions. The Probability of Error (Pe) is defined as:

$$P_e = \frac{\text{number of bits in error}}{\text{number of bits transmitted}}$$

Although the test definition is simple, it is necessary to ensure that the test conditions are fully specified when comparing performance curves. The signal-to-noise ratio (SNR), is a measure of received signal power to received noise power. Care must be taken when comparing performance curves to ensure that the bandwidth of the noise used in both cases is identical. Another key parameter is the power of the received signal.



a. **Bias Distortion**



b. **Isochronous Distortion**

Figure 5.4 Bias and Isochronous Distortions

Two of the most common formats for presenting Bit Error Rate (BER) data are indicated in Figure 5.5. Figure 5.5a illustrates a plot in which receive level is fixed, and SNR varies, while Figure 5.5b illustrates a plot in which SNR is fixed and receive level is varied. Both types of plots are required to fully understand a modem's performance. For instance, Figure 5.5a tends to indicate that Modem "A" is superior, whereas Figure 5.5b reveals that, at low signal levels, Modem "B" is in fact superior.

5.5 TELEPHONE NETWORK IMPAIRMENTS

There are several different types of impairments that can be expected on the telephone network. It is often desirable to measure and plot a modem's Bit Error Rate performance in the presence of these imperfections.

5.5.1 Telephone Line Type

As discussed, modem performance is affected by the amplitude and group delay distortion of the telephone line. It is common for both modem manufacturers and users to measure modem performance under varying amplitude and group delay conditions.

5.5.2 Carrier Frequency Offset

Frequency shifts may occur on the public switch telephone network due to the up- and down-conversion of individual telephone calls to higher frequencies by the network. This conversion is necessary to allow multiple simultaneous channels on single cables or microwave links; however, a small shift in the frequency of the modem carrier may result.

5.5.3 Phase Jitter

Phase jitter is a cyclical change of the phase of a carrier due to imperfections of the amplification and repeating equipment present on the telephone network. It generally occurs at multiples of the power supply frequency. Instruments to simulate phase jitter are available, allowing the user to vary the frequency and peak-to-peak value of the jitter.

5.5.4 Impulse Noise

The noise generally used for BER plots is "white" noise, containing components of all frequencies. However, it is also common for the telephone network to contain "impulse" noise, consisting of short bursts or clicks. Instruments to simulate these bursts are available, allowing the user to vary the rise time, duration, magnitude, and frequency of occurrence of the noise.

5.5.5 Hits

In addition to impulse noise, a telephone channel may exhibit abrupt positive or negative changes in its phase or amplitude characteristics. Instruments to simulate these "hits" are also available. The significant parameters are the same as for impulse noise.

5.5.6 Harmonics

The amplifiers and repeaters present on telephone lines may not be completely linear. This results in harmonics of the transmitted signal being added to the signal itself.

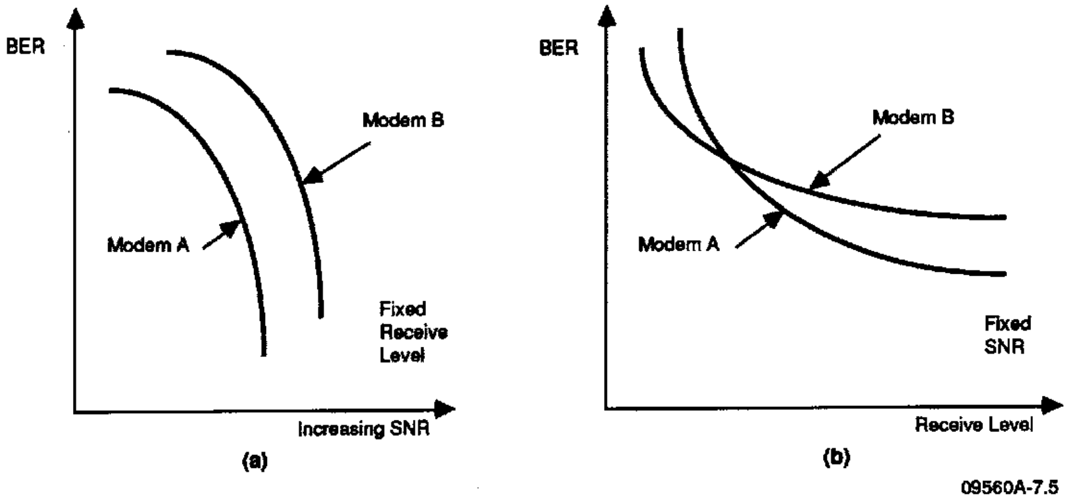


Figure 5.5 BER Data Plots

APPENDIX

Abbreviated Product Specifications

Am79101

WORLD-CHIP™
Autodial FSK Modem

DISTINCTIVE CHARACTERISTICS

- Bell 103, 113, 108 and CCITT V.21 compatible at 300 bps full-duplex
- Bell 202 and CCITT V.23 compatible at 1200 bps half-duplex with up to 150 bps back channel (CCITT V.23 modes with optional soft carrier turn-off feature)
- Single-chip Digital Signal Processor
- Autodial support
  - Dual-Tone Multi-Frequency (DTMF) tone generation
  - Call Progress Tones Detection
  - Answerback Tones Detection
- Integral 4- to 2-wire hybrid
- Public Switched Telephone Network (PSTN) response times
- Serial RS-232C/CCITT V.24 type handshake interface and protocol

GENERAL DESCRIPTION

The Am79101 World Chip is a single-chip asynchronous Frequency Shift Keying (FSK) modem that is compatible with the applicable Bell and CCITT-recommended standards for 103/113/108, 202, V.21 and V.23-type modems. All modulation, demodulation, filtering, analog-to-digital and digital-to-analog functions are provided on-chip.

Using the features described below, an intelligent autodial, auto answer FSK modem may be implemented with only an Am79101 single-chip under the control of a host microprocessor and a Protective Phone Line Interface (PPLI) circuit.

The modem operates in a serial asynchronous mode, the serial interface supports the RS-232C/CCITT V.24-type handshake signals at TTL levels.

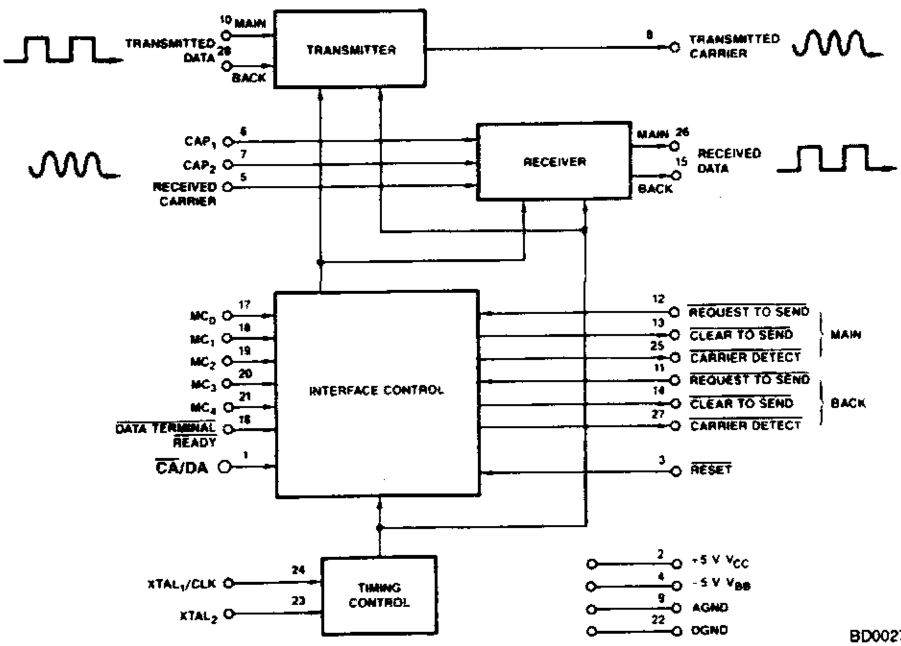
The modem analog interface provides an internal hybrid for the 4- to 2-wire conversion. Auxiliary functions performed within the Am79101 include:

Autodial support with DTMF generation and Call Progress Tones Detection.
Discrete answer tones detection (Bell and CCITT).
Autoanswer support.
Analog loopback support.

The Am79101 is housed in 28-pin plastic leaded chip carrier and 28-pin plastic and ceramic dual-in-line packages.

Connection to the telephone network may be via a PPLI or an acoustic coupler. All digital I/O signals are TTL-compatible (except the external clock and $\overline{\text{RESET}}$ signals) and the circuit operates from ±5 volts.

BLOCK DIAGRAM



BD002791

Am7910

WORLD-CHIP™
FSK Modem

DISTINCTIVE CHARACTERISTICS

- Complete FSK Modem in a 28-pin package – just add line interface
- Compatible with Bell 103/113/108, Bell 202, CCITT V.21, CCITT V.23 specifications
  - 1200 bps full duplex on 4-wire line
- All digital signal processing, digital filters, and ADC/DAC included on-chip
  - No external filtering required
- Includes essential RS-232/CCITT V.24 handshake signals
  - Dial-up network response times
- Autoanswer capability
- Local copy/test modes

GENERAL DESCRIPTION

The Am7910 is a single-chip asynchronous Frequency Shift Keying (FSK) voiceband modem. It is pin-selectable for baud rates of 300, 600 or 1200 bits per second, and is compatible with the applicable Bell and CCITT-recommended standards for 103/113/108, 202, V.21 and V.23-type modems. Five mode control lines select a desired modem configuration.

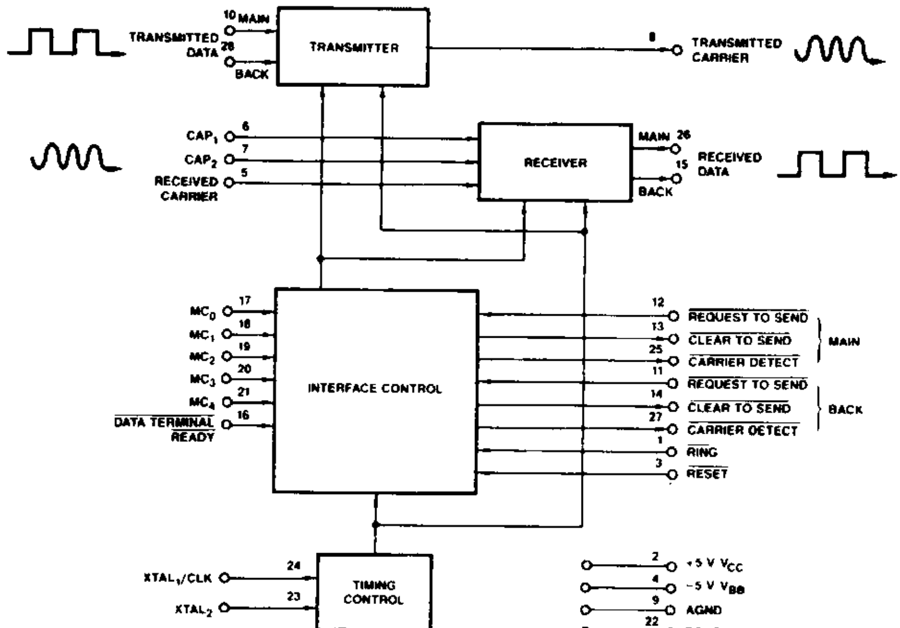
Digital signal processing techniques are employed in the Am7910 FSK Modem to perform all major functions such as modulation, demodulation, and filtering. The Am7910 contains on-chip analog-to-digital and digital-to-analog converter circuits to minimize the external components in a system. This device includes the essential RS-232/CCITT V.24 terminal control signals with TTL levels.

Clocking can be generated by attaching a crystal to drive the internal crystal oscillator or by applying an external clock signal.

A Data Access Arrangement (DAA) or acoustic coupler must provide the phone line interface externally.

The Am7910 is fabricated using N-channel MOS technology in a 28-pin package. All the digital input and output signals (except the external clock and RESET signals) are TTL-compatible. Power supply requirements are ±5 volts.

BLOCK DIAGRAM



BD002791

Am7911

WORLD-CHIP™
FSK Modem

DISTINCTIVE CHARACTERISTICS

- Complete FSK Modem in a 28-pin package – just add line interface
- Compatible with Bell 103/113/108, Bell 202, CCITT V.21, CCITT V.23 specifications
  - –Mode-selectable 5 or 150 baud back channel for Bell 202
  - –Up to 150 baud on V.23 back channel
  - –CCITT V.23 modes with optional soft carrier turn-off feature
  - –1200 bps full duplex on 4-wire line
- All digital signal processing, digital filters, and ADC/DAC included on-chip
  - –No external filtering required
- Includes essential RS-232/CCITT V.24 handshake signals
  - –Fast response times for leased-line networks
- Autoanswer capability
- Local copy/test modes

GENERAL DESCRIPTION

The Am7911 is a single-chip asynchronous Frequency Shift Keying (FSK) voiceband modem intended for use in leased-line applications. It is pin-selectable for baud rates of 300, 600, or 1200 bits per second, and is compatible with the applicable Bell and CCITT recommended standards for 103/113/108, 202, V.21 and V.23 type modems. Five mode control lines select a desired modem configuration; for Bell 202 applications, the Am7911 modem provides a 150 bps back channel in addition to the standard 5 bps channel, and for V.23 up to 150 bps back channel operation is possible.

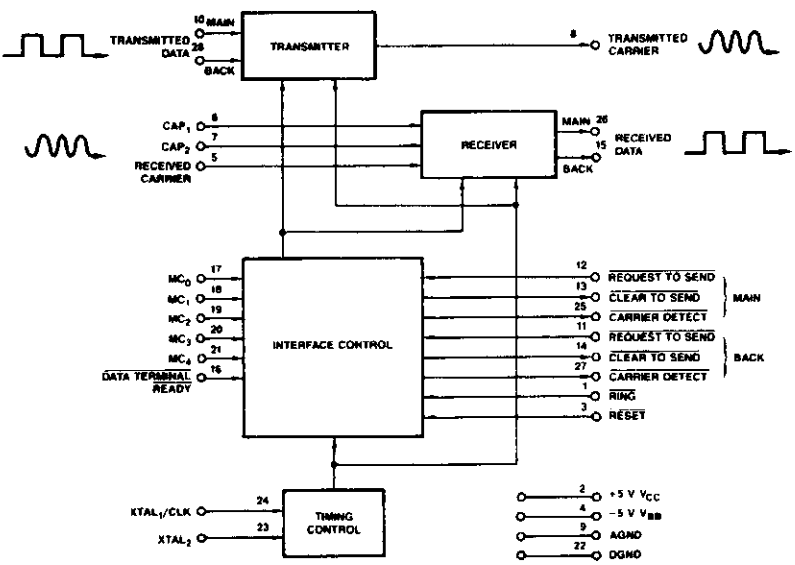
Digital signal processing techniques are employed in the Am7911 modem to perform all major functions such as modulation, demodulation, and filtering. The Am7911 contains on-chip analog-to-digital and digital-to-analog converter circuits to minimize the external components in a system. This device includes the essential RS-232/CCITT V.24 terminal control signals with TTL levels.

Clocking can be generated by attaching a crystal to the internal crystal oscillator amplifier or by applying an external clock signal.

A Data Access Arrangement (DAA) or acoustic coupler must provide the phone line interface externally.

The Am7911 is fabricated using N-channel MOS technology in a 28-pin package. All the digital input and output signals (except the external clock and $\overline{\text{RESET}}$ signals) are TTL compatible. Power supply requirements are ±5 volts.

BLOCK DIAGRAM



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