SD indicator. If the SD indicator flashes, this indicates that data from the attached terminal is reaching the modem and it also verifies that there is continuity on pin 2 of the cable connecting the terminal and modem.

### TR indicator

The terminal ready (TR) indicator on the Courier 2400 modem reflects the state of the terminal device's data terminal ready (DTR) control signal. The TR indicator is thus labeled DTR on the T1000 modem as well as on many other modems.

When the TR indicator is illuminated this denotes that the terminal or computer port is ready to establish communications. Similarly, a lack of illumination indicates that the attached terminal or computer port is not ready to establish communications. You can easily observe the latter state by powering on your modem prior to powering on an attached personal computer. Only after the computer is powered on and you have loaded communications software and prepared your program to go on-line will the TR indicator illuminate.

### MR indicator

The modem ready (MR) indicator illuminates when the Courier 2400 modem is powered on. When you first power on your modem the MR indicator should illuminate to inform you that the device is operational.

### AL indicator

The analog loopback (AL) indicator is illuminated when your modem is in an analog loopback mode of operation. When your modem is placed in this mode of operation data transmitted from the attached terminal device is sent to the modem's transmitter where it is modulated and looped back to the modem's receiver. The receiver demodulates the data and sends it back to the attached terminal device where you can visually verify the operation of the modem.

When the AL indicator is illuminated the modem is not connected to the telephone line and cannot transmit data. Thus, you must enter the appropriate command to place the modem back into its data mode to communicate with another modem.

### Other indicators

Three other common modem indicators that warrant discussion are CS or CTS, ARQ, and SYN.

The CS or CTS indicator denotes the state of the RS-232 clear to send control signal. When the indicator is illuminated, this fact tells you that the modem has received a valid carrier tone and is ready to modulate data. The ARQ (automatic repeat request) indicator illuminates when the modem is in its MNP error detection and correction mode, having successfully made an MNP connection with a remote modem. The SYN (synchronous) indicator is applicable to modems that can

operate synchronously. This indicator illuminates when the modem is placed into its synchronous mode of operation.

## Modem testing

To assist users in isolating problems, most modems include a built-in diagnostic testing capability. This diagnostic testing capability ranges in scope from a simple self-test of the modem's circuitry to several more sophisticated types of loopback tests.

### Self-test

Most modems include a self-test capability that is either initiated as a stand-alone test or is used in conjunction with a loopback test. The basic modem self-test is designed to verify the operation of the modem's internal circuitry and uses a pattern generator to produce a known sequence of data which will be used by the transmitter.

Figure 4.42 illustrates the operation of a basic modem self-test. Once initiated, the modem's transmitter is tied to its receiver breaking any prior connection to a communications line. Data modulated by the transmitter is demodulated by the receiver and passed to a pattern comparator which consists of circuitry as well as a section of ROM that contains the same data sequence which is generated by the pattern generator. The pattern comparator thus allows the demodulated data to be compared to the sequence generated when the self-test is initiated. If the received data does not match the contents of the pattern comparator the modem's circuitry will cause a status indicator, normally labeled ER for error, to illuminate. This action informs you that the modem is defective and should be returned to the manufacturer.

The basic self-test previously described was originally limited to inclusion in leased line modems. This was because most switched network modems can be easily attached to terminals or personal computers which can be used as a pattern generator. The operator's vision can then be used as a pattern comparator. Today, a growing percentage of switched network modems include a self-testing capability that can be used with one or more loopback modes of operation to test a local or remote modem or the communications path between modems.
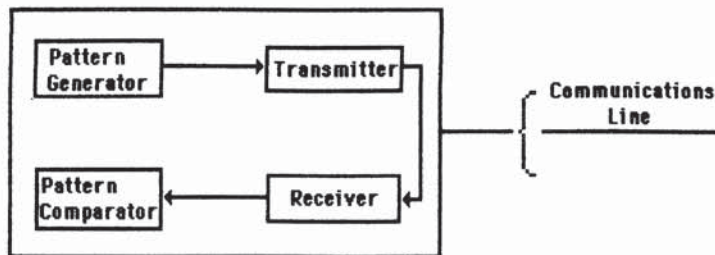


**Figure 4.42**   Modem self-test

## Loopback tests

There are four types of loopback tests modems may be able to perform. Each of these tests is illustrated in Figure 4.43. Readers should note that the loopback testing capability of modems varies both by manufacturer as well as by a manufacturer's product line. Although most leased line modems will be capable of initiating all four loopback tests illustrated in Figure 4.43, many switched network modems are limited to performing a lesser number of loopback tests.
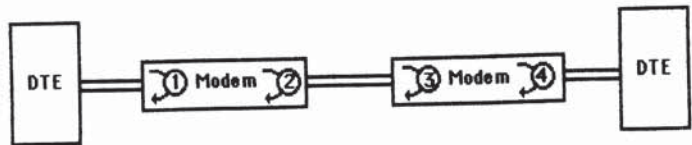


**Figure 4.43** Modem loopbacks: 1 local digital loop, 2 local analog loop, 3 remote analog loop, 4 remote digital loop

The local digital loop test illustrated in Figure 4.43 is employed to test the operation of the data terminal equiment (DTE) to include determining if data is leaving the terminal or computer port. In comparison, the local analog loop tests the digital and analog circuits of the modem to determine if they are in working condition. The remote analog loop is used for testing the line to the remote modem, while the remote digital loop is used to check the operating condition of the line and the local and remote modems. Now that we have an overview of the use of each loopback test, let us examine them in more detail.

### Local digital loopback

The initiation of a digital loopback test causes the local modem to tie its transmitter and receiver sections together, in effect, bypassing its modulator and demodulator. The input data received from a connected DTE on pin 2 of an RS-232 conductor are thus routed to pin 3 which is the receive data conductor. In effect, a local digital loop results in the establishment of two loops as illustrated in Figure 4.44.
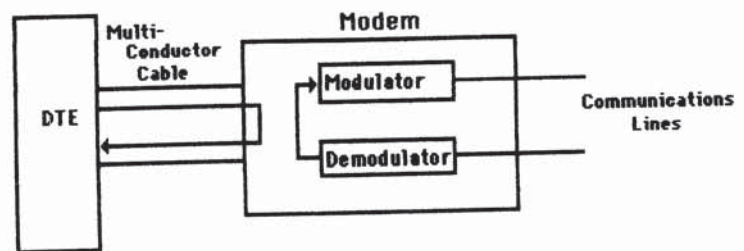


**Figure 4.44** Local digital loop test

A local digital loop can be used to test the operation of the DTE as well as verify the multiconductor cable connecting the DTE to the modem. To accomplish those functions you would first initiate a local digital loop.

Once this is accomplished, you can type on the keyboard of your terminal or personal computer and visually observe the characters echoed back to your terminal or PC. After you are satisfied that your DTE and cable are performing correctly you can terminate the loopback.

### Local analog loopback

The local analog loopback test is designed to verify the operation of the analog circuits of a modem, including its modulator and demodulator. During this test two loops are established, resulting in the modem's modulator being looped to its receiver, while the send and receive paths of the communications line are tied together. The local analog loop test is illustrated in Figure 4.45.
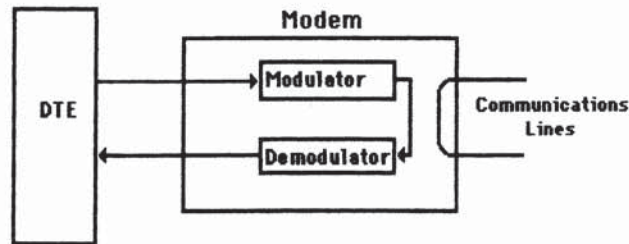


**Figure 4.45**   Local analog loop test

### Remote analog loopback

The remote analog loopback results in the remote modem bridging the transmit and receive wire pairs of a leased line together. This allows you to use test equipment at the local site to determine the quality of the line. Since modems designed for use on the PSTN operate on a two-wire system this loopback test is not included in switched network modems.

### Remote digital loopback

The remote digital loopback test results in the remote modem demodulating received data and then passing it to its transmitter for remodulation back onto the line. This test is similar to the local digital loopback test previously illustrated in Figure 4.44. Once the remote modem is placed into its digital loopback mode of operation data transmitted from the local site is looped at the remote modem back to the data originator, in effect, providing a mechanism for checking both the quality of the circuit as well as the operation of the local and remote modems.

While analog and digital loopbacks are the main tests built into modems, several vendors offer additional diagnostic capabilities that may warrant attention. A few of

these diagnostic tests that deserve mention include bit error rate testing and alarm threshold monitoring. Normally these two diagnostic functions are implemented by the user obtaining a network control system (NCS) designed to operate with modems manufactured by the company that produced the NCS. Typically, the NCS is a microcomputer-based system that monitors the status of the modems in an organization's network, generates alarms when certain predefined conditions occur, and in some cases it actually performs corrective action prior to a disruption in network operation occurring.

The NCS is normally installed at the central computer site and on a periodic basis queries the status of the remote modems on both point-to-point and multidrop lines. Through the 1980s most NCS systems worked in conjunction with modems that perform testing and status queries via the use of a secondary channel. This permits data transmission to continue on the primary channel, with testing and responses to the testing flowing concurrently at a lower data rate on the secondary channel. Since then NCS systems were developed to work in conjunction with intelligent modems designed to recognize certain datapatterns as a request to perform testing and/or issue status responses. When used with this type of modem, one's actual information flow of data is interrupted by the testing.

To allow users to implement network control functions in a data communications network without having to replace existing modems, some vendors offer what is commonly called a 'modem wraparound unit' or modem network adapter. Such devices are cabled to both ends of the modem and in schematic diagrams appear to wraparound the modem, hence the term 'wraparound unit'. Figure 4.46 illustrates the utilization of an NCS manufactured by one vendor with local and remotely located modems manufactured by a second vendor by the employment of two modem wraparound units. Through the use of the wraparound units, the NCS can conduct modem and line tests, even when the modems are manufactured by different vendors.
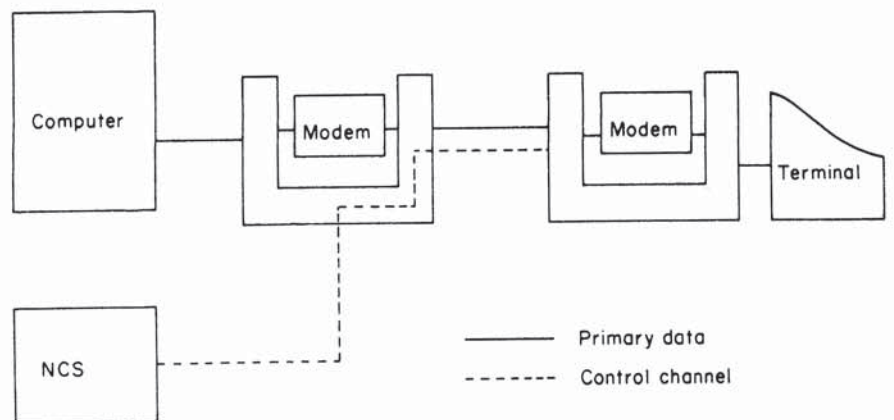


**Figure 4.46** Modem wraparound unit utilization. The modem wraparound unit permits the use of a network control station produced by one vendor with modems manufactured by other vendors

## 4.3 INTELLIGENT MODEMS

In this section we will examine how microprocessors and read only memory (ROM), random access memory (RAM), and erasable programmable read only memory (EPROM) are used to add intelligence to modems. Although the ability to add intelligence to modems can provide an almost infinite number of features and functions, we will focus our attention upon three key areas that are normally associated with the term 'intelligent modem'—command sets, error detection and correction, and data compression.

The command set used by a modem governs its operational capability, enabling automatic dialing and redialing of telephone numbers, selection of an operating transmission rate, and other functions to be performed via communications software or by a person typing commands from his or her terminal or personal computer to the modem.

An error detection and correction capability results in the transmitting modem adding a special 'trailer' to each block of data which is formed according to a predefined algorithm. The receiving modem performs the same algorithm on each received data block and compares its computation to the transmitted 'trailer'. If the transmitted and computed 'trailers' match, the data block is considered to be received correctly. If they do not match, the receiving modem will transmit this fact to the transmitting modem, which will then retransmit the block. The correction of data thus occurs by retransmission.

When an intelligent modem performs data compression it examines data for redundancy, replacing, as an example, strings of characters by a special code and count which the receiving modem recognises and uses to regenerate the data back into its original form. Through the use of data compression a modem's effective information transfer capability is increased since each character transmitted may represent more than one data character.

### Command sets

During the 1970s, a small communications company called Bizcom developed the concept of adding intelligence to modems through the use of codes that the modem would recognize and act upon. The series of codes that a modem is designed to interpret and act upon is referred to as its command set.

In addition to the pioneering work of Bizcom, other modem manufacturers developed similar performing modems. Unfortunately, during the early days of intelligent modem developments the command sets built into modems manufactured by different vendors were more than likely to be incompatible with one another. Many communications software products were thus originally restricted to operating with a specific modem or a few modems based upon the command sets the software supported. Due to the popularity of the Hayes Microcomputer Products series of Smartmodems™ which at one time captured over 50% of the personal computer modem market, that firm's command set emerged as a *de facto* industry standard. During the 1980s many modem manufacturers incorporated a similar command set into their products, resulting in a high level of compatibility

between communications software that supports the command set used by Hayes Smartmodems and modems manufactured by other vendors.

Today, many modem manufacturers advertise their products as 'Hayes compatible'; however, this term can be misleading as it may not indicate whether or not 'Hayes compatible' software will support all of the features of a particular modem. In addition, communications software advertised as supporting the Hayes command set may not support all of the features of a specific Hayes Microcomputer Products Smartmodem. To understand why these incompatibilities can occur and how they can be alleviated, let us first examine the Hayes command set and then focus our attention upon several modem products and the command sets they support.

## The Hayes command set

The Hayes command set actually consists of a basic set of commands and command extensions. The basic commands, such as placing the modem off-hook, dialing a number and performing similar operations are common to all Hayes modems. The command extensions, such as placing a modem into a specific operating speed are only applicable to modems built to transmit and receive data at that speed.

The commands in the Hayes command set are initiated by transmitting an attention code to the modem, followed by the appropriate command or set of commands one desires the modem to implement. The attention code is the character sequence AT, which must be specified as all uppercase or all lowercase letters. The requirement to prefix all command lines with the code AT has resulted in many modem manufacturers denoting their modems as Hayes AT compatible.

The command buffer in a Hayes Smartmodem holds 40 characters, permitting a sequence of commands to be transmitted to the modem on one command line. This 40-character limit does not include the attention code, nor does it include spaces included in a command line to make the line more readable. Table 4.21 lists the major commands included in the basic Hayes command set. Other modems, such as Telebit's Trailblazer employ a command set that can be considered to be a superset of the Hayes command set. While communications software that uses the Hayes command set will operate with this modem, such software cannot utilize the full potential of the modem. This is because the Hayes command set does not support several Telebit Trailblazer features, such as its proprietary method of error detection.

The basic format required to transmit commands to a Hayes compatible intelligent modem is shown below.

> AT Command[Parameter(s)]Command[Parameter(s)] ... Return

Each command line includes the prefix AT, followed by the appropriate command and the command's parameters. The command parameters are usually the digits 0 or 1, which serve to define a specific command state. As an example, H0 is the command that tells the modem to hang up or disconnect a call, while H1 is the command that results in the modem going off-hook, which is the term used to define the action that occurs when the telephone handset is lifted. Since

**Table 4.21** Basic Hayes command set (major commands)

| Command | Description |
|---------|-------------|
| AT | Attention |
| A | Answer call |
| A/ | Repeat last command |
| B | Select the method of modem modulation |
| C* | Turn modem's carrier on or off |
| D | Dial a telephone number |
| E | Enable or inhibit echo of characters to the screen |
| F* | Switch between half- and full-duplex modem operation |
| H | Hang up telephone (on hook) or pick up telephone (off hook) |
| I | Request identification code or request check sum |
| L | Select the speaker volume |
| M | Turn speaker off or on |
| O | Place modem on-line |
| P | Pulse dial |
| Q | Request modem to send or inhibit sending of result code |
| R* | Change modem mode to 'originate-only' |
| S | Set modem register values |
| T | Touch-tone dial |
| V | Send result codes as digits or words |
| X | Use basic or extended result code set |
| Y | Enable or inhibit long space disconnect |
| Z | Reset the modem |
| + + + | Escape command |

Commands followed by an asterisk (*) are not applicable to the Hayes V-series Smartmodem 2400 but are recognized by earlier Hayes modems and modems manufactured by many other vendors.

many commands do not have parameters, those terms are enclosed in brackets to illustrate that they are optional. A number of commands can be included in one command line as long as the number of characters does not exceed 40, which is the size of the modem's command buffer. Finally, each command line must be terminated by a carriage return character.

## Command use

To illustrate the utilization of the Hayes command set let us assume we desire to automatically dial New York City information. First, we must tell the modem to go off-hook, which is similar to one manually picking up the telephone handset. Then we must tell the modem the type of telephone system we are using, pulse or touch-tone, and the telephone number to dial. Thus, if we have a terminal or personal computer connected to a Hayes compatible modem, we would send the following commands to the modem:

AT H1
AT DT1,212-555-1212

In the first command, the 1 parameter used with the H command places the modem off-hook. In the second command, DT tells the modem to dial (D) a

telephone number using touch-tone (T) dialing. The digit 1 was included in the telephone number since it was assumed we have to dial long-distance, while the comma between the long-distance access number (1) and the area code (212) causes the modem to pause for 2 s prior to dialing the area code. This 2 s pause is usually of sufficient duration to permit the long-distance dial tone to be received prior to dialing the area code number.

Since a Smartmodem automatically goes off-hook when dialing a number, the first command line is not actually required and is normally used for receiving calls. In the second command line, the type of dialing does not have to be specified if a previous call was made, since the modem will then use the last type specified. Although users with only pulse dialing availability must specify P in the dialing command when using a Hayes Smartmodem, several vendors now offer modems that can automatically determine the type of dialing facility the modem is connected to and then use the appropriate dialing method without requiring the user to specify the type of dialing. For other non-Hayes modems, when the method of dialing is unspecified, such modems will automatically attempt to perform a touch-tone dial and, if unsuccessful, then redial using pulse dialing.

To obtain an appreciation of the versatility of operations that the Hayes command set provides, assume two personal computer users are communicating with one another. If the users wish to switch from modem to voice operations without hanging up or redialing, one user would send a message via the communications program he or she is using to the other user indicating that voice communications is desired. Then, both users would lift their telephone handsets and type + + + (Return) ATH (Return) to switch from on-line operations to command mode (hang-up). This will cause the modems to hang-up, turning off the modem carrier signals and permitting the users to converse.

## Result codes

The response of the Smartmodem to commands is known as result codes. The Q command with a parameter of 1 is used to enable result codes to be sent from the modem in response to the execution of command lines while a parameter of 0 inhibits the modem from responding to the execution of each command line.

If the result codes are enabled, the V command can be used to determine the format of the result codes. When the V command is used with a parameter of 0, the result codes will be transmitted as digits, while the use of a parameter of 1 will cause the modem to transmit the result codes as words. Table 4.22 lists the Basic Results Codes set of the Hayes Smartmodem 2400. As an example of the use of these result codes, let us assume the following commands were sent to the modem:

    AT Q0
    AT V1

The first command, AT Q0, would cause the modem to respond to commands by transmitting result codes after each command line is executed. The second command, AT V1, would cause the modem to transmit each result code as a word code. Returning to Table 4.22 this would cause the modem to generate the word

**Table 4.22** Smartmodem 2400 result codes

| Digit code | Word code |
| --- | --- |
| 0 | OK |
| 1 | CONNECT |
| 2 | RING |
| 3 | NO CARRIER |
| 4 | ERROR |
| 5 | CONNECT 1200 |
| 6 | NO DIALTONE |
| 7 | BUSY |
| 8 | NO ANSWER (replaces NO CARRIER if the @ is included in the dial string) |
| 9 | Reserved for future use |
| 10 | CONNECT 2400 |

code 'CONNECT' when a carrier signal is detected. If the command AT V0 was sent to the modem, a result code of 1 would be transmitted by the modem, since the 0 parameter would cause the modem to transmit result codes as digits.

By combining an examination of the result codes issued by a Smartmodem with the generation of appropriate commands, software can be developed to perform such operations as redialing a previously dialed telephone number to resume transmission in the event a communications session is interrupted and automatically answering incoming calls when a ring signal is detected.

## Extended AT commands

A large number of extended AT commands were developed by Hayes Microcomputer Products to control the operation and utilization of special modem features. Table 3.21 lists the extended AT commands supported by the Hayes V-series Smartmodem 2400. Note that each of these commands uses the ampersand (&) character to indicate it is an extension to the basic command set.

Hayes Smartmodems operating above 2400 bps support most, if not all, of the extended commands listed in Table 4.23 while adding certain extended commands that reflect the addition of certain features beyond those included in the V-series Smartmodem 2400. For example, a Hayes Smartmodem 9600 that supports the V.32 standard includes the extended commands &U0 and &U1. The &U0 extended command enables trellis coding when the modem operates at 9600 bps, while the &U1 extended command disables trellis coding.

All extended AT commands are initiated in the same manner as basic commands and have the same rules and constraints with respect to their use. For example, the &C command must be prefixed by 'AT' or 'at' to be recognized by the modem. Similarly, the modem must be in an off-line or an on-line command mode to be able to recognize and act upon the command. You should not confuse the Hayes V-series Smartmodem 2400 extended AT commands listed in Table 4.23 with the term 'extended AT commands' many modem vendors advertise. The latter is

**Table 4.23** Enhanced AT command set (Hayes V-series Smartmodem 2400)

| Command | Description |
| --- | --- |
| &C0 | Assume data carrier always present |
| &C1 | Track presence of data carrier |
| &D0 | Ignore DTR signal |
| &D1 | Assume command state when an on-to-off transition of DTR occurs |
| &D2 | Hang up and assume command state when an on-to-off transition of DTR occurs |
| &D3 | Reset when an on-to-off transition of DTR occurs |
| &D4 | Reset when enter low power mode when DTR is low |
| &G0 | No guard tone |
| &G1 | 550 Hz guard tone |
| &G2 | 1800 Hz guard tone |
| &J0 | RJ-11/RJ-41S/RJ-45S telco jack |
| &J1 | RJ-12/RJ-13 telco jack |
| &K0 | Disable local flow control |
| &K3 | Enable RTS/CTS local flow control |
| &K4 | Enable XON/XOFF local flow control |
| &K5 | Enable transparent XON/XOFF local flow control |
| &M0 | Asynchronous mode |
| &M1 | Synchronous mode 1 |
| &M2 | Synchronous mode 2 |
| &M3 | Synchronous mode 3 |
| &R0 | Track CTS according to RTS |
| &R1 | Ignore RTS; always assume presence of CTS |
| &S0 | Assume presence of DSR signal |
| &S1 | Track presence of DSR signal |
| &T0 | Terminate test in progress |
| &T1 | Initiate local analog loopback |
| &T3 | Initiate local digital loopback |
| &T4 | Grant request from remote modem for RDL |
| &T5 | Deny request from remote modem for RDL |
| &T6 | Initiate remote digital loopback |
| &T7 | Initiate remote digital loopback with self test |
| &T8 | Initiate local analog loopback with self test |
| &W0 | Save storable parameters of active configuration as profile 0 |
| &W1 | Save storable parameters of active configuration as profile 1 |
| &X0 | Modem provides transmit clock signal |
| &X1 | Data terminal provides transmit clock signal |
| &X2 | Receive carrier provides transmit clock signal |
| &Y0 | Specify stored user profile 0 as power-up configuration |
| &Y1 | Specify stored user profile 1 as power-up configuration |
| &Z | Store telephone number |

usually a term used by vendors other than Hayes Microcomputer Products to define the commands used by their products to access and control such features as MNP, V.42, V.42 bis, and proprietary methods of error control and data compression. Microcom has defined a series of AT command extensions to control and access error control, data compression, and flow control. Used by the Microcom AX series of modems, the extensions have been adopted by Intel, Okidata, Practical Peripherals, Prometheus, and Ven-Tel. Modems manufactured by other vendors

normally use unique commands and S register settings to access and control those previously mentioned features.

## Modem registers

A third key to the degree of compatibility between non-Hayes and Hayes Smartmodems is the number, use and programmability of registers contained in the modem. Hayes Smartmodems contain a series of programmable registers that govern the function of the modem and the operation of some of the commands in the modem's command set. Table 4.24 lists the functions of the first 13 registers built into the Hayes Smartmodem 2400, to include the default value of each register and the range of settings permitted. These registers are known as S registers, since they are set with the S command in the Hayes command set. In addition, the current value of each register can be read under program control, permitting software developers to market communications programs that permit the user to easily modify the default values of the modem's S registers.

**Table 4.24** S register control parameters

| Register | Function | Default value | Range |
|----------|----------|---------------|-------|
| S0 | Ring to answer on | * | 0–255 |
| S1 | Counts number of rings | 0 | 0–255 |
| S2 | Escape code character | ASC11 43 | ASC11 0–127 |
| S3 | Carriage return character | ASC11 13 | ASC11 0–127 |
| S4 | Line feed character | ASC11 10 | ASC11 0–127 |
| S5 | Backspace character | ASC11 8 | ASC11 0–127 |
| S6 | Dial tone wait time (s) | 2 | 2–255 |
| S7 | Carrier wait time (s) | 30 | 1–255 |
| S8 | Pause time caused by comma (s) | 2 | 0–255 |
| S9 | Carrier detect response time (1/10s) | 6 | 1–255 |
| S10 | Time delay between loss of carrier and hang-up (1/10s) | 7 | 1–255 |
| S11 | Touch-tone duration and spacing time (ms) | 70 | 50–255 |
| S12 | Escape sequence guard time in units of 20 ms | 50 | 0–255 |

*Usually depends on a switch setting

To understand the utility of the ability to read and reset the values of the modem's S registers, consider the time period a Smartmodem waits for a dial tone prior to going off-hook and dialing a telephone number. Since the dial tone wait time is controlled by the S6 register, a program offering the user the ability to change this wait time might first read and display the setting of this register during the program's initialization. The reading of the S6 register would be accomplished by the program sending the following command to the modem:

AT S6?

The modem's response to this command would be a value between 2 and 255, indicating the time period in seconds that the modem will wait for a dial tone. Assuming the user desires to change the waiting period, the communications program would then transmit the following command to the modem, where $n$ would be a value between 2 and 255.

AT S6 = $n$

## Compatibility

For a non-Hayes modem to be fully compatible with a Hayes modem, command-set compatibility, result-codes compatibility and modem-register compatibility is required. Of the three, the modem-register compatibility is usually the least important and many users may prefer to consider only command-set and response-codes compatibility when acquiring intelligent modems.

The rationale for omitting register compatibility from consideration is the fact that many non-Hayes modem vendors manufacture 'compatible' modems using the default values of the Hayes Smartmodem registers. This enables those manufacturers to avoid building the S registers into their modems, reducing the size, complexity and often also reducing the price of their modem. Thus, if the default values of the S registers are sufficient for the user and the modem under consideration is both command-set and result-code compatible the issue of register compatibility can normally be eliminated as an acquisition factor to consider.

A second rationale for omitting register compatibility from consideration is the fact that other non-Hayes modem vendors incorporate 100 or more registers into their products, usually with their first 13 being compatible to the first 13 Hayes S registers. The remaining registers are used mainly to control such advanced modem features as flow control, protocol spoofing, data compression, and error control.

Of the first 13 S registers perhaps the most critical to understand is the S7 register. This register is used by an originating modem to set the time that it will wait to receive a carrier tone from a distant modem prior to hanging up. For example, assuming you access a 9600 bps modem with a 1200 or 2400 bps modem, the setting of the S7 register will normally govern whether or not you will be able to establish communications. To understand the criticality of the S7 register it is important to understand how the remote modem operates upon receiving a call.

When a remote 9600 bps modem receives a call it first attempts to negotiate an operating rate. In response to the incoming call it responds with a carrier tone at a predefined frequency. If it does not receive a recognized response within approximately 15 s and supports multiple modulation schemes that are enabled the remote modem then transmits a second carrier tone which represents a lower operating rate. This process continues until the remote modem either receives a recognized carrier tone from the originating modem or the seting of the remote modem's S7 register is reached. If the latter occurs, the remote modem will disconnect the incoming call. While most data center and packet network dial-in ports have their modem S7 registers set to a minimum of 60 s, most modem users fail to note that when their modem is used to originate calls it may not receive a recognizable carrier

tone for 45 or more seconds when accessing a higher-speed modem. Thus, you should normally change the factory default setting of 30 s of the S7 register to a higher value when dialing a high-speed modem with a Bell System 212A or V.22 bis modem. The preceding is also true if you use a V.32 modem set to operate at 9600 bps to access a V.34 modem set to operate at 28 800 bps, or whenever there are more than three operating rate differences between calling and answering modem.

## Error detection and correction

Error detection and correction, also known as error control and error correction, has been implemented in a large number of switched network modems using a variety of techniques. Each technique follows a common methodology in that data transmitted by an attached terminal device formally known as data terminal equipment or DTE is first gathered into a block of characters. Next, an algorithm is applied to the block to generate one or more checksum characters that are appended to the block for transmission. The receiving modem performs the same algorithm on the block it receives, less its checksum character or characters. The checksum computed by the receiving modem is referred to as the locally generated checksum, which is compared to the transmitted checksum. If both the locally generated and transmitted checksums are equal, the data block is assumed to have been received error-free. Otherwise, the data block is assumed to have one or more bits in error, and the receiving modem then requests the transmitting modem to retransmit the data block. Thus, error correction is accomplished by retransmission.

Since error correction is accomplished by retransmission, all modems that perform error detection and correction must have buffers to temporarily store data until an acknowledgement from the receiving modem occurs, at which time the transmitting modem can discard the stored data block that was received error-free by the modem at the opposite end of the transmission path. Since modem data buffers are finite in size, a mechanism is required to control the flow of data from the attached terminal to the modem. This mechanism is known as flow control. As all modems that perform error detection and correction must perform flow control, we will first examine this subject area prior to investigating the methods by which error detection and correction are performed by modems.

## Flow control

Flow control is a mechanism used to compensate for the difference between the rate data that reaches a device and the rate at which a device processes and transmits data. To illustrate the concept behind flow control, consider Figure 4.47 which illustrates its rationale. Assume this modem has an operating rate of 28 800 bps. If the compression ratio of the modem is 2:1, on the average, every two characters of data entering the modem will be compressed into one character. Thus, the terminal device can be configured to transfer data to the modem at 57 600 bps which is twice the data rate at which the modem transfers data onto the PSTN.
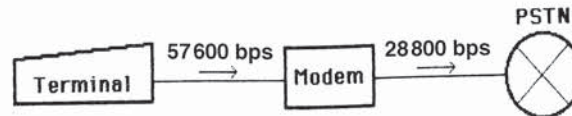
**Figure 4.47** Rationale for flow control. When the modem achieves a 2:1 compression ratio, data entering the device at 57 600 bps is immediately placed on the line at 28 800 bps. When the compression ratio is under 2:1 data cannot exit the modem as fast as it enters the modem and is placed into a buffer area. To prevent the buffer from overflowing, the modem requires a mechanism to stop the terminal from sending additional data—a process known as flow control

As long as the modem can compress two characters into one, the terminal device can keep on transferring data to the modem at 57 600 bps. Suppose, however, a portion of the data entering the modem cannot be compressed or is compressible at a ratio less than 2:1. Either situation would cause data flowing into the modem to be lost. To prevent a potential data loss, modems that compress data or perform other functions, including error detection and correction, will include data buffers as temporary storage locations to compensate for the difference between the data flow into the modem and the rate at which the modem can process and transmit data.

Since data buffers represent a finite amount of storage, a modem that accepts data at one rate and transfers it at a different rate must be able to control the flow of data into its buffers. Known as flow control, this technique is used to prevent modem buffers from overflowing and losing data.

To illustrate how modem flow control operates, consider Figure 4.48 which illustrates a modem's buffer storage area. As data enters the modem from an attached terminal device at a faster rate than it can be placed on the line, the buffer begins to fill. When the occupancy of the buffer reaches a predefined high level ($H_L$) the modem initiates flow control to inhibit additional data from flowing into its buffer. Otherwise, the buffer could continue to fill until data eventually cannot be stored and is then lost. As data flows out of the buffer and is modulated and placed on the line, the point of low occupancy is reached ($L_L$). When this level of occupancy is reached the modem uses flow control to enable transmission to resume to the modem. Now that we have reviewed why flow control is used in modems, let us examine the methods by which it is implemented.
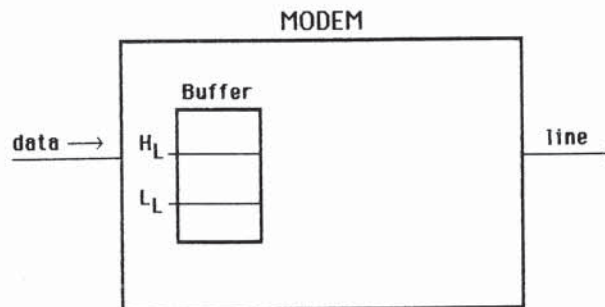


**Figure 4.48** Flow control and buffer utilization. $H_L$ = high level to disable data flow, $L_L$ = low level to enable data flow

## Methods of flow control

There are three primary methods by which flow control can be implemented, including the use of RTS/CTS control signals and the transmission of the character pairs XON/XOFF and ENQ/ACK. In addition, some devices support the use of a mixture of flow control methods.

### RTS/CTS signaling

When a modem receives a RTS (request to send) signal from a terminal device and is ready to receive data, it will respond by raising its CTS (clear to send) signal. One method to control the flow of data into the modem is thus for the modem to turn off its CTS signal whenever it wants to stop the flow of data into its buffers. Whenever the modem wants the flow of data to resume it will then raise or turn on its CTS signal. This method of flow control is referred to as hardware flow control as the controlling signals at the interface govern the operation of the hardware.

### XON/XOFF

The use of XON and XOFF characters for flow control is referred to as in-band signaling as the characters flow over pin 3 (receive data) to the attached terminal device. In comparison, the use of the CTS control signal which is not a data signal is referred to as out-band signaling.

Many asynchronous terminal devices recognize an ASCII code of 19, which represents the CTRL-S character, as a signal to suspend transmission. This XOFF signal is also known as a DC3 (device control number 3) character and is issued by the modem when it wants an attached DTE to suspend the flow of data to the modem. Once the modem has emptied its buffer to a low level it will transmit an XON character to the DTE, which serves as a signal for the DTE to resume transmission. The XON character has an ASCII code of 17 and represents the CTRL-Q character. This character is also known as the DC1 (device control number 1) character.

One of the problems associated with the use of XON/XOFF flow control is the use of those characters in a data file by an application program which, when transmitted, can cause a DTE to be infinitely suspended. For example, suppose you are transmitting a WordStar file to another computer system. That word processing program uses DC3s generated by a CTRL-S as markers for the beginning and end of underscoring. If your modem is set to echo characters and you have DC3s in a file, the first DC3 modulated and transmitted to the distant system is also echoed back to the attached terminal device. The DTE detects the DC3 as an XOFF request and correctly responds by suspending transmission and waiting for an XON from the modem which will never come. Here the obvious solution to the problem is to turn echo off for the file transfer. Unfortunately, this obvious solution is not obvious to many modem users, especially when they are transmitting and receiving binary files. For this reason it is highly recommended that you use hardware flow control if you intend to transfer binary files.

### ENQ/ACK

The enquire/acknowledge method of flow control is used with certain Hewlett Packard computers and terminal devices. This method of flow control is based upon the DTE sending an enquire (ENQ) and then receiving an acknowledgement (ACK) prior to actually transmitting data. Under the Hewlett-Packard ENQ/ACK protocol the DTE that receives an ACK in response to its ENQ transmits a block of data of variable size that can be up to approximately 2000 characters in length.

When a modem is configured to use ENQ/ACK flow control it will respond to the DTE's ENQ with an ACK when it can accept data. At that time it will release the ENQ to flow to the remote modem.

### DTE flow control

The previously described methods of flow control are also applicable in the reverse direction, with DTEs controlling the flow of data.

DTEs are similar to modems that contain buffers, in that when operating communications software, a finite area of storage is reserved for receiving data to be processed. If the received data is transferred to a peripheral device, such as a printer operating slower than the communications line, this situation could result in a loss of data. This is because the modem would be passing demodulated data to the DTE faster than it could empty its buffer area, eventually resulting in a loss of data when the buffer is full. Like modems, DTEs capable of performing flow control may be able to support three methods to accomplish this task: RTS/CTS signaling and the use of the character pairs XON/XOFF and ENQ/ACK.

Under RTS/CTS signaling the DTE will drop its RTS signal as an indicator that it wants to suspend receiving data. In a full-duplex operation the modem will stop sending data to the DTE when RTS drops and resume transmission when the RTS signal is raised. Under half-duplex operations the modem will transmit the contents of its buffer to the DTE and drop its CTS signal in response to the DTE lowering its RTS signal.

When XON/XOFF flow control is used by the DTE the local modem will suspend sending data to that device when it receives an XOFF. Upon receipt of an XON the modem will resume sending data to the DTE.

As previously mentioned, some Hewlett-Packard computers and terminals support the ENQ/ACK protocol. When both modems are configured to use the ENQ/ACK protocol the receipt of an ENQ from a remote DTE is passed through both modems to the local DTE. Until the local DTE responds with an ACK the local modem will not send data to that device.

Now that we have examined the methods by which modems perform flow control let us focus our attention upon the methods by which error detection and correction can be accomplished.

## Methods of error detection and correction

Until 1989 error detection and correction was conspicuous by the absence of a *de jure* standard. Several error detection and correction techniques were developed

based upon the use of cyclic or polynomial code error detection schemes previously described in Chapter 1. After a modem computes a CRC it is appended to the block of data to be transmitted. The receiving modem uses the same predefined generating polynomial to generate its own CRC based upon the received block and then compares the locally generated CRC with the transmitted CRC. If the two match, the receiving modem transmits a positive acknowledgement to the transmitting modem, which not only informs the distant modem that the data was received correctly but can also serve to inform the remote modem that if additional blocks of date remain to be transmitted the next block can be sent. If an error has occurred, the locally generated CRC will not match the transmitted CRC and the receiving modem will transmit a negative acknowledgement which informs the remote modem to retransmit the previously transmitted data block.

Although most modem manufacturers used the CRC-16 polynomial, $X^{16}+X^{15}+X^5+1$, which has the bit composition 1100000000010001, to operate against each data block, incompatibilities between the methods used to block data and transmit negative and positive acknowledgements resulted in the method of error detection and correction employed by one vendor being incompatible with the method used by another modem manufacturer. The one major exception to this incompatibility between vendor error detection and correction methods is the Microcom Networking Protocol (MNP), which has been licensed by Microcom to a large number of modem manufacturers. Until 1989, MNP was considered as a *de facto* standard due to a base of approximately one million modems supporting one or more MNP classes. In 1989 the MNP method of error detection and correction was recognized by the ITU as one of two methods for performing this function when the V.42 recommendation was promulgated.

## Rationale

One of the issues that confuse many modem users is the rationale for using a modem's error detection and correction feature. After all, file transfer protocols such as XMODEM, YMODEM, ZMODEM, and their derivatives also provide an error detection and correction capability.

When a modem's error control feature is enabled and operates successfully in conjunction with a distant modem, error detection and correction is operating during the entire communications session. This means that regardless of the function you are performing, whether reading an electronic mail message, transferring a file, or sending a message to SYSOP, your transmission is protected. In comparison, the error detection and correction function embedded into a file transfer protocol operates only during the file transfer. Thus, a logical question you may have is why you should use a file transfer protocol with error detection and correction when your modem performs that function.

Although you need to use a file transfer protocol to transfer a file in an orderly manner, the error detection and correction function of the file transfer protocol may be redundant in most situations, because the file transfer protocol protects data from computer to computer, while the modem's error detection and correction feature protects data from modem to modem. Thus, the former protocol protects data leaving the serial port and flowing to the modem. However, since data errors

due to cabling are extremely rare, in most cases the double protection from a file transfer protocol and error correcting modems between modems can actually cause transmission delays and is not necessary. Some modems incorporate a protocol spoofing feature, the use of which eliminates the transmission delays associated with the use of error detection and correction file transfer protocol with error correction modems. If the modem you are using does not support a suitable form of protocol spoofing, you should consider the use of a 'streaming file transfer protocol' to transfer files when using an error correcting modem. This term refers to a protocol that does not perform any error checking and whose use improves the transfer speed of information. An example of a streaming file transfer protocol is YMODEM-G. Since streaming file transfer protocols do not preform error detection and correction, they should only be used with modems operating in their error correction mode of operation.

## MNP

The Microcom Networking Protocol (MNP) was developed by the modem manufacturer Microcom, Inc., to provide a sophisticated level of error detection and correction as well as to enhance the data file transfer of intelligent modems. Microcom has licensed their MNP for use by other modem vendors, resulting in a large number of manufacturers incorporating this protocol into their products. Today, the MNP error-correcting protocol is considered a *de facto* industry standard, with over ten million modems having this feature when this book was published.

The MNP protocol was designed in a layered fashion like the OSI reference model developed by the International Organization for Standardization. MNP contains three layers instead of the seven layers in the OSI reference model. Figure 4.49 illustrates the correspondence between the OSI reference model and the MNP protocol.

The MNP link layer is responsible for establishing a connection between two devices. Included in the link layer is a set of negotiations that are conducted between devices to enable them to agree upon such factors as the transmission mode (full-or half-duplex), how many data messages can be transmitted prior to requiring a confirmation and how much data can be obtained in a single message. After these values are established the link layer initiates the data transfer process as well as performing error detection and correction through the use of a frame checking scheme.

Figure 4.50 illustrates the format of an MNP frame of information which has similarities to both bisynchronous and HDLC communications. Each frame contains three bytes which act as a 'start flag'. The SYN character tells the receiver that a message is about to arrive, the combination of data link escape (DLE) and start of text (STX) informs the receiver that everything following is part of the message. The first header describes the user data, such as the duplex setting, number of data messages before confirmation, etc. The session header defines additional information about the transmitted data which enables the automatic negotiation of the level of service that can be used between devices communicating with one another. Currently there are ten versions or classes of the MNP protocol

OSI reference model                                    MNP protocol
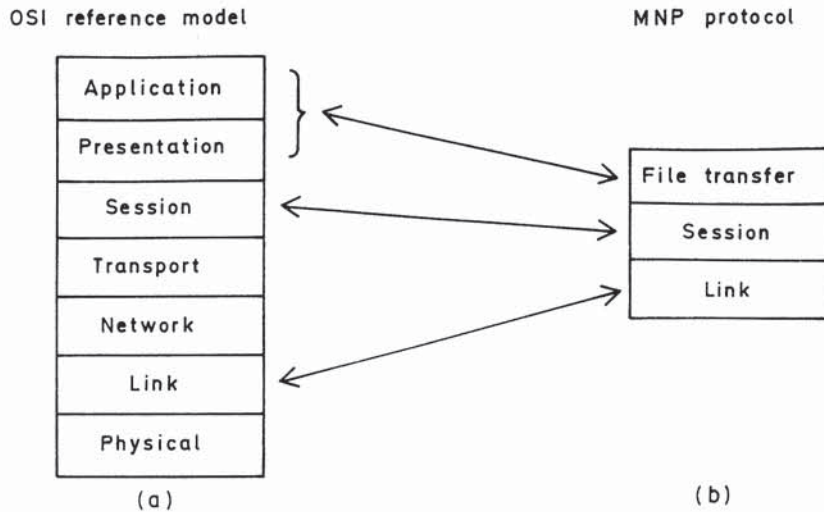


**Figure 4.49**   OSI reference model and MNP protocol: comparison
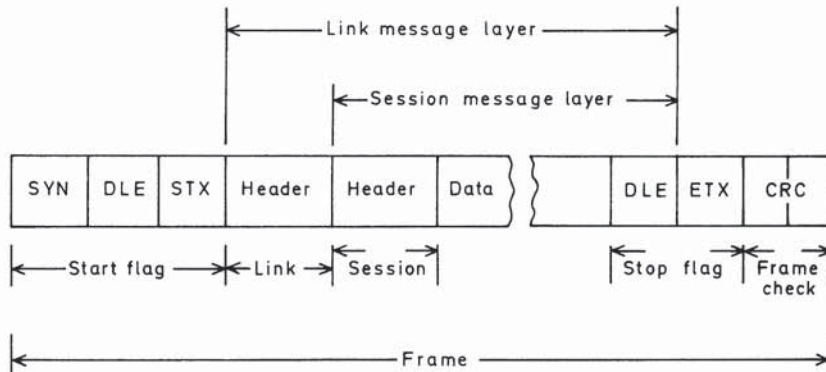


**Figure 4.50**   MNP block format

(see Table 4.25), with each higher level adding more sophistication and efficiency. When an MNP link is established the protocol assumes that the devices on both sides can only operate at the lowest level. Then, the devices negotiate with each other to determine the highest mutually supported class of MNP services they can support. If a non-MNP device is encountered the MNP device reverts to a 'dumb' operating mode, providing an MNP modem with the ability to be used with non-MNP devices.

The error correction capability of MNP actually occurs under its Class 4 operation. Modems that are MNP error control compatible will thus be advertised as MNP Class 4 compliant. Under Class 4, the actual framing of data depends upon whether the data is asynchronous or synchronous. If the attached terminal device transmits asynchronous data, the MNP frame format is as indicated in

**Table 4.25** MNP protocol classes

| Protocol class | Description |
| --- | --- |
| Class 1 | The lowest performance level, uses an asynchronous byte-oriented half-duplex method of exchanging data. The protocol efficiency of a Class 1 implementation is about 70% (a 2400 bps modem using MNP Class 1 will have a 1690 bps throughput). |
| Class 2 | Uses asynchronous byte-oriented full-duplex data exchange. The protocol efficiency of a Class 2 modem is about 84% (a 2400 bps modem will realize 2000 bps throughput). |
| Class 3 | Uses synchronous bit-oriented full-duplex data exchange. This approach is more efficient than the asynchronous, byte-oriented approach, which takes 10 bits to represent 8 data bits because of the 'start' and 'stop' framing bits. The synchronous data format eliminates the need for start and stop bits. Users still send data asynchronously to a Class 3 modem, but the modems communicate with each other synchronously. The protocol efficiency of a Class 3 implementation is about 108% (a 2400 bps modem will actually run at a 2600 bps throughput). |
| Class 4 | Adds two techniques—Adaptive Packet Assembly and Data Phase Optimization. In the former technique, if the data channel is relatively error-free, MNP assembles larger data packets to increase throughput. If the data channel is introducing many errors, then MNP assembles smaller data packets for transmission. Although smaller data packets increase protocol overhead, they concurrently decrease the throughput penalty of data retransmissions—more data are successfully transmitted on the first try. Data phase optimization is a technique for eliminating some of the administrative information in the data packets, which further reduces protocol overhead. The protocol efficiency of a Class 4 implementation is about 120% (a 2400 bps modem will effectively yield a throughput of 2900 bps). |
| Class 5 | This class adds data compression, which uses a real-time adaptive algorithm to compress data. The real-time capabilities of the algorithm allow the data compression to operate on interactive terminal data as well as file transfer data. The adaptive nature of the algorithm refers to its ability to continuously analyze user data and adjust the compression parameters to maximize data throughput. The effectiveness of data compression algorithms depends on the data pattern being processed. Most data patterns will benefit from data compression, with performance advantages typically ranging from 1.3 to 1.0 and 2.0 to 1.0, although some files may be compressed at an even higher ratio. Based on a 1.6 to 1 compression ratio, Microcom gives Class 5 MNP a 200% protocol efficiency, or 4800 bps throughput in a 2400 bps modem installation |
| Class 6 | This class adds 9600 bps V.29 modulation, universal link negotiation, and statistical duplexing to MNP Class 5 features. Universal link negotiation allows two unlike MNP Class 6 modems to find the highest operating speed (between 300 and 9600 bps) at which both can operate. The modems begin to talk at a common lower speed and automatically 'negotiate' the use of progressively higher speeds. Statistical duplexing is a technique for simulating full-duplex service over half-duplex high-speed carriers. Once the modem link has been established using full-duplex V.22 modulation, user data streams move via the carrier's faster half-duplex mode. The modems, however, monitor the data streams and allocate each modem's use of the line to best approximate a full-duplex exchange. Microcom claims that a 9600 bps V.29 modem using MNP Class 6 (and Class 5 data compression) can achieve 19.2 kbps throughput over dial circuits. |

**Table 4.25**   (*continued*)

| Protocol class | Description |
| --- | --- |
| Class 7 | Uses an advanced form of Huffman encoding in conjunction with a predictive Markov algorithm to represent user data in the shortest possible Huffman codes. This compression technique is called *Enhanced Data Compression*. Enhanced Data Compression has all of the characteristics of Class 5 compression, but, in addition, predicts the probability of repetitive characters in the data stream. Class 7 compression, on the average, reduces data by 42%. In addition to Class 5 and Class 7 data compression, modern implementation of MNP now supports the ITU-T V.42 bis data compression standard |
| Class 8 | Adds ITU V.29 fast-train modem technology to Class 7 Enhanced Data Compression, enabling half-duplex devices to emulate full-duplex transmission. |
| Class 9 | Combines V.32 modem modulation technology with Class 7 Enhanced Data Compression, resulting in full-duplex throughput that can exceed that obtainable with a V.32 modem by 300%. Class 9 also employs *selective retransmission*, in which error packets are retransmitted, and *piggybacking*, in which acknowledgement information is added to the data. Retransmission is facilitated by indicating in the error or Negative Acknowledgement (NAK) packet the order sequence of each of the failed messages. Instead of having to retransmit all messages from the point of error to include the good messages, only the failed messages are resent. |
| Class 10 | Adds Adverse Channel Enhancement (ACE), which optimizes modem performance in environments with poor or varying conditions, such as cellular communications, rural telephone service, and some international connections. Adverse Channel Enhancements fall into five categories: <ul><li>Negotiated Speed Upshift: modem handshake begins at lowest possible modulation speed, and when line conditions permit, the modem upshifts to the highest possible speed.</li><li>Robust Auto-Reliable Mode: enables MNP10 modems to establish a reliable connection during noisy call setups by making multiple attempts to overcome circuit interference. In comparison, other MNP classes make only one call setup attempt.</li><li>Dynamic Speed Shift: causes an MNP10 modem to adjust its operating rate continuously throughout a session in response to current line conditions. This adjustment includes both downshifting and upshifting across modulation schemes, which enables modems to use the highest possible operating rate a given line connection can support.</li><li>Aggressive Adaptive Packet Assembly: results in packet sizes varying from 8 to 256 bytes in length. Small data packets are used during the establishment of a link, and there is an aggressive increase in the size of packets as conditions permit. The computations required under Aggressive Adaptive Packet Assembly to determine the optimum packet size are performed during idle time.</li><li>Dynamic Transmit Level Adjustment (DTLA): designed for cellular operations, DTLA results in the sampling of the modem's transmit level and its automatic adjustment to optimize data throughput. Once a cellular connection is established, transmit-level statistics are continually sampled using link management packets. When necessary, transmit levels are adjusted based on the existing conditions to optimize data throughput.</li></ul> |

byte #

| | byte # |
|---|---|
| SYN Character | 1 |
| DLE Character | 2 |
| STX Character | 3 |
| Header | 4 |
| Information | : |
| DLE Character | N-3 |
| ETX Character | N-2 |
| Frame Check Sequence | N-1 / N |

(a)

| | byte # |
|---|---|
| Flag | 1 |
| Header | 2 |
| Information | : |
| Frame Check Sequence | N-2 / N-1 |
| Flag | N |

(b)
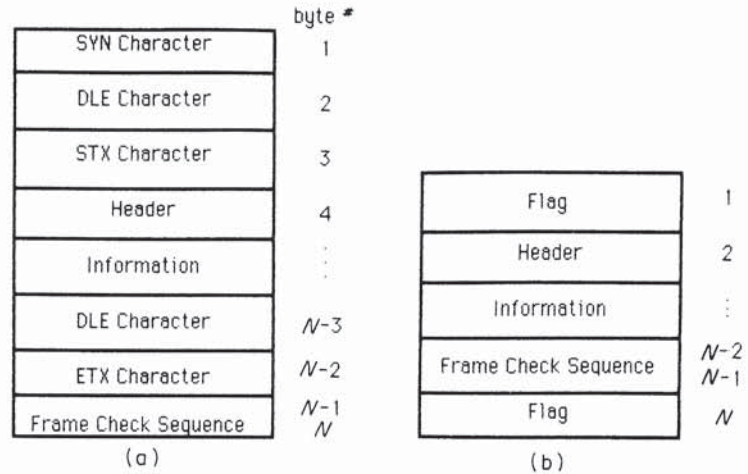
**Figure 4.51** MNP frame format (DLE data link escape, STX start of text, ETX end of text): (a) Asynchronous data frame format; (b) Synchronous data frame format

Figure 4.51(a). If the attached terminal device transmits synchronous data, the frame format is as indicated in Figure 4.51(b). For both frame formats the frame check sequence (FCS) characters are generated by a CRC-16 polynomial.

Under MNP a special frame called a link request (LR) that has a predefined format is used to establish an error-corrected connection between two modems. This frame is also used to negotiate a variety of parameters between two MNP modems, including the maximum number of frames that can be outstanding and not acknowledged at a particular point in time, the maximum information field length and the framing mode to be used. The link request frame is developed by using a value of one in the header field in the synchronous frame format illustrated in Figure 4.51(b). If during negotiation the framing mode is negotiated for the transmission of asynchronous data the frame format illustrated in Figure 4.51(a) will be used.

One of the more interesting aspects of MNP is its Class 10, which is designed as an enabling technology architecture. MNP Class 10 can be viewed as residing on top of different modulation standards, such as V.22 bis, V.32, V.32 bis, V.34, and other standards as well as the *de jure* and *de facto* standards for data compression and error control. This enables an MNP Class 10 modem to provide users with maximum performance while assuring compatibility with all recognized international standards. Figure 4.52 illustrates the MNP Class 10 architecture.

## MNP PDUs

From a technical perspective, an MNP frame's header and information field are called a Protocol Data Unit (PDU). The PDU provides the information required to define and negotiate facilities, manage communications, and transfer user data between modems. Eight types of PDUs are currently defined. Each has the general

| Enhanced Capability | MNP Class 10 | | | | | |
|---|---|---|---|---|---|---|
| Data Compression | V.42bis | | | MNP 5/7 | | |
| Error Control | LAPM | | | MNP2-4 | | |
| Modulation | 103A | 212A | V.22bis | V.32 | V.32bis | V.34 |

**Figure 4.52**　MNP Class 10 architecture

format:

| LEN | TYPE | PARAMETERS... |
|---|---|---|

where LEN represents the length of the PDU in bytes, TYPE is a numeric code of 1 through 9 that indicates the type of PDU, while PARAMETERS represent a string of octets that have special meaning. Both the position of the parameter in the PDU and the code in the TYPE octet are used to define the role of the parameter. With the ability to include up to 256 parameters in a PDU and define up to 256 different PDUs, MNP can be considerably expanded. In fact, it is very easy to enhance MNP by adding new parameters to PDUs or adding new PDUs since compatibility with existing MNP implementations is maintained as the use of new enhancements must be mutually agreed on.

Figure 4.53 illustrates the basic formats of the eight MNP PDUs which are transported within an asynchronous or synchronous frame. Both the length and type fields in each PDU are one octet in length. Up to 256 different parameters can be carried within a PDU and user data is attached to the LT PDU as a variable length field up to 256 bytes in length.

The Link Request PDU is used to initiate or acknowledge a connection request. This PDU can transport such parameters as the MNP service class, the number of messages that must be acknowledged, and the maximum amount of user data that will be carried in a message.

The Link Disconnect PDU is used to request that the link should be disconnected. Information in the parameters field indicates the reason for the disconnection request.

The Link Transfer PDU is used to move user data across an established connection. User data is attached to this PDU as a variable length field of up to 256 bytes. The only parameter that is included in this PDU is the Send Sequence Number (SSN) which identifies the location of the current PDU within a stream of PDUs being transmitted.

The Link Acknowledgement PDU is used to acknowledge the receipt of a message. Parameters within the PDU indicate how many additional Link Transfer

Link Request (TYPE=1 'for LR')

| LEN | LR | PARAMETERS . . . |

initiates or acknowledges
a connection request.

Link Disconnect (TYPE=2 'for LD')

| LEN | LD | PARAMETERS . . . |

Link Transfer (TYPE=4 'for LT')

| LEN | LT | PARAMETER . . . |

Link Acknowledgement (TYPE=5 'for LA')

| LEN | LA | PARAMETERS . . . |

Link Attention (TYPE=6 'for LN')

| LEN | LN | PARAMETERS . . . |

Link Attention Acknowledgement (TYPE=7 'for LNA')

| LEN | LNA | PARAMETERS . . . | user data |

Link Management (TYPE=8 'for LM')

| LEN | LM | PARAMETERS . . . |

Link Management Acknowledgement (TYPE=9 'for LMA')

| LEN | LMA | PARAMETERS . . . |

**Figure 4.53**   MNP ADUs

(LT) PDUs the receiver can accept prior to it becoming overwhelmed. Thus, this PDU can be viewed as a mechanism which controls a sliding window of PDUs.

The Link Attention PDU is used to interrupt communications between the transmitting and receiving device, such as when a PC user initiates a break signal. The parameters within this PDU indicate whether the transmitted data that was sent but not yet delivered should be delivered prior to the message interrupt, after the interrupt, or discarded.

The Link Attention Acknowledgement PDU is used by a receiving device to acknowledge the receipt of a previously transmitted Link Attention (LN) PDU. Among the parameters included in this PDU is the SSN of the LN PDU being acknowledged.

The Link Management PDU is used to implement MNP Class 6 and Class 10 functions. Thus, this PDU would be used to direct a change in the operating rate of a modem during a communications session.

The last PDU presently defined, Link Management Acknowledgement, acknowledges receipt of a Link Management PDU. Thus the LMA PDU would include the SSN of the LMA being acknowledged.

### MNP session

An MNP session involves the exchange of PDUs as illustrated by the time chart in Figure 4.54. Communication is initiated when one modem transmits a Link Request PDU. The receiving modem responds with its own Link Request PDU permitting the two to agree upon a set of operating parameters when the originator returns its Link Acknowledgement PDU. This three-way handshaking process is referred to as the MNP call establishment process.

Once a call is established the originator transmits data using Link Transfer PDUs. The receiver acknowledges a specific number of Link Transfer PDUs with a Link Acknowledgement PDU, with the number of PDUs based upon negotiations
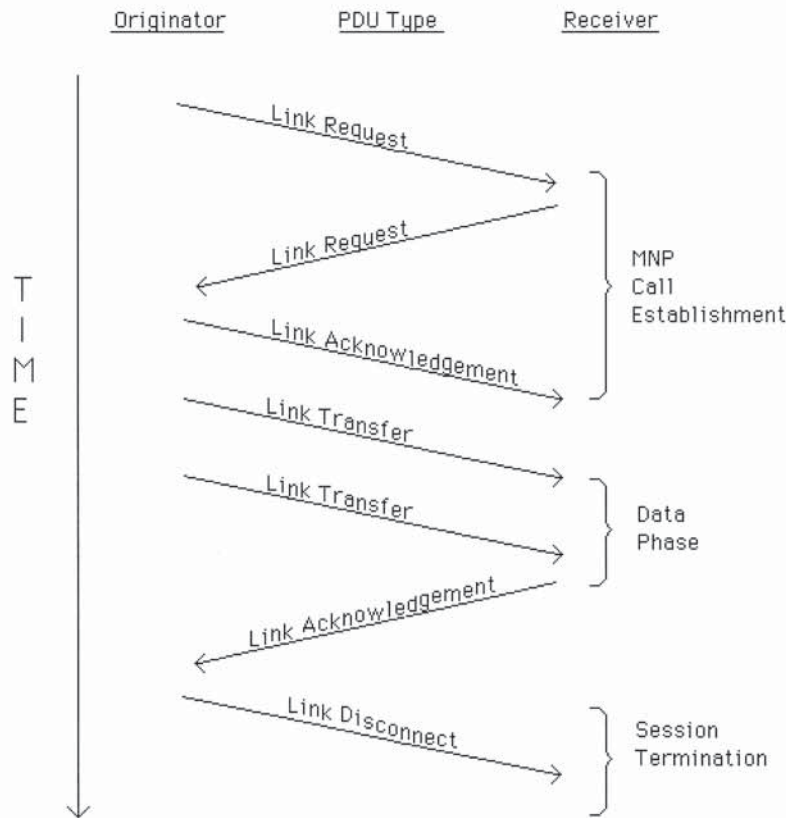
**Figure 4.54**  Simple MNP session

that previously occurred in the exchange of Link Requests during the call establishment process. The exchange of user data and acknowledgements is known as the data phase.

At any time either modem can issue a Link Disconnect PDU. Once issued, the sender assumes the link is terminated. Similarly, the receiver of the LD PDU also considers the connection terminated upon receipt of the sender's request to disconnect.

If an error should occur, a modem will transmit a Link Acknowledgement PDU. That PDU will tell the originator to transmit LT PDUs following the one whose SSN is specified in the Link Acknowledgement parameter. Thus, error control, while not indicated in Figure 4.54, represents a relatively simple process under MNP.

### Cellular enhancements

One major problem that occurs on cellular transmission that is much more pronounced than when transmission occurs on land lines is noise. This is especially true when a user is really mobile, transmitting from a moving vehicle. As the vehicle moves, the laptop or notebook modem constantly changes its physical (spatial) relationship with cellular base towers. What might represent bothersome noise on a voice call may result in an excess number of retransmissions that result in the communications protocol terminating a session.

Although MNP-10 provides a reasonable level of cellular transmission capability, when used in a mobile environment transmission at data rates above 9.6 kbps are very difficult to obtain. In addition, MNP requires two compatible Class 10 modems to enhance cellular transmission capability.

Recognizing the limitations of MNP-10, Rockwell Semiconductor developed an extension to MNP-10 known as MNP-10EC, with the suffix initials standing for Enhanced Cellular. MNP-10EC can be considered as a new and improved version of MNP-10 for which Rockwell adapted the Physical Layer so that it is 'cellular aware', in which a signal blank condition occurring when moving between cell sites or when a signal is obscured by a building is recognized as a temporary condition. This enables the modem to recover the signal when it reappears without terminating the connection.

The actual operation of MNP-10EC is at the Protocol layer, with that layer combined with the cellular-aware Physical layer in Rockwell cellular modems. The Protocol layer is responsible for an adjustment of the modem's operating rate and packet size based upon signal quality and error rates. Thus, both signal blanking and signal quality impairments are compensated for by MNP-10EC. According to Rockwell, operating rates up to 14.4 kbps are achievable through the use of its MNP extension.

### Enhanced Throughput Cellular

A competitive cellular-specific protocol to MNP-10EC is known as Enhanced Throughput Cellular (ETC). Developed by AT&T's formally owned Paradyne unit, this protocol represents a proprietary extension to V.32 bis and V.42

technologies. An ETC-capable modem can change its operating rate every five seconds to accommodate fluctuating levels of line noise, signal strength, and jitter.

The key to performance of an ETC modem is the manner by which it transmits frames. The modem uses 32-byte frames and permits up to 15 frames to be outstanding prior to requiring an acknowledgement.

While both MNP-10EC and ETC represent important developments for achieving higher operating rates on cellular facilities, another often overlooked method for improving transmission is the connection of a modem to a cellular telephone. Most cellular modems are cabled to a cellular telephone which results in a small amount of signal loss. One exception to this is a cellular direct-connect feature recently introduced by Megahertz Corporation. This modem was manufactured to connect directly to the cellular telephone without requiring the use of a cable. At the time this book was written Megahertz Corporation's Modem PC Card with a cellular direct-connect capability was capable of being used with certain Motorola and Nokia cellular phones.

## LAP-M

As previously mentioned, the MNP error detection and correction method is one of two methods recognized under the ITU V.42 recommendation. In actuality, the MNP method was recognized as an alternative procedure under V.42, with the primary method known as link access protocol-modem or LAP-M.

Under V.42 the originating modem transmits an originator detection pattern (ODP). The ODP is defined as the bit sequence

$$01000\ 10001\ 11\ldots11\ 01000\ 10011\ 11\ldots11$$

which represents the DC1 character with even parity, followed by 8 to 16 ones, followed by DC1 with odd parity, followed by 8 to 16 ones. A V.42 compatible modem will respond to the ODP with an answer detection pattern (ADP), whose bit format will indicate that V.42 is supported or that no error-correcting protocol is desired. If the ADP is not observed within a predefined period of time, for which V.42 modems use a default value of 750 ms, it is assumed that the distant modem does not possess V.42 error-correcting capability. In this situation the originating modem may fall back to a non-error-correcting mode of operation, or if it incorporates MNP, it can then attempt to negotiate an MNP error detection and correction mode of operation.

Under the LAP-M protocol a different frame structure from MNP is used. In addition, the frame check sequence characters are generated by a different polynomial. Thus, MNP and LAP-M are completely incompatible with each other.

## Compatibility issues

Since V.42 specifies MNP as an alternative procedure it is essentially an option that modem manufacturers may or may not include in a V.42 modem. Thus, if you wish

to communicate using the MNP method of error detection and correction you can either consider purchasing a modem that is MNP Class 4 compatible or which is V.42 compatible and supports the alternative error-correction procedure. Fortunately, most modem vendors that have adopted V.42 support also support the alternative error-correction procedure more commonly referred to as MNP Class 4.

## Data compression

Similar to error detection and correction, modem manufacturers have incorporated a variety of data compression algorithms into their products. Until 1989, most compression methods, while representing a proprietary scheme of one vendor, were licensed by that vendor to other manufacturers. This resulted in several *de facto* data compression standards being incorporated into modems manufactured by different vendors.

In 1990 the CCITT promulgated the V.42 bis recommendation which defines a new data compression method known as Lempel-Ziv as an international standard. Unlike V.42, which concerns error detection and correction and specifies MNP Class 4 as an alternative, V.42 bis does not specify an alternative method of data compression.

### Rationale

A data compression performing modem results in each modulated bit conveying more information than an equivalent modem that does not compress data. To illustrate this concept assume that over a period of time the ratio of characters flowing into a modem's compression circuitry to the number of compressed characters output and modulated by the modem is 2:1. This 2:1 compression ratio indicates that, on average, a compression performing modem can transfer information twice as fast as a similar modem that does not compress data.

Modem vendors have incorporated data compression into their products to provide users with two key advantages over non-compression performing modems —a reduction in transmission time and an alternative to purchasing higher-speed and higher-cost modems. The reduction in transmission time is probably the more obvious of the two advantages as compacted data takes less time to be transmitted. Concerning the latter advantage, consider a data compression modem that operates at 14 400 bps and has a 2:1 compression ratio. This modem would provide you with an effective throughput of 28 800 bps and would probably be less expensive than a 28 800 bps modem since the modulation scheme required to operate at the higher data rate is more complex and more costly than the modulation method used by 14 400 bps modems.

### MNP Class 5 compression

Of the two methods of data compression supported by the Microcom Networking Protocol, Class 5 is the more popular as it predates Class 7. Since each MNP Class

is downward negotiable for compatibility, a modem that supports MNP Class 7 will communicate with a modem that is MNP Class 5 compatible using MNP Class 5 data compression.

MNP Class 5 specifies that the sending modem apply two modifications to the transmitted data stream in an attempt to reduce the number of bits actually sent. The first manipulation or data compression method is referred to as run-length encoding, while the second method of compression is known as adaptive frequency encoding.

MNP Class 5 uses run-length encoding to avoid sending long sequences of repeated data octets, where each octet represents eight bits that can define a character in a particular character set or any binary value from 0 to 255 represented by the individual bit settings within the octet. The MNP Class 5 version of run-length encoding results in a repetition count being inserted into the data stream to represent the number of repeated data octets which follow the first three occurrences of a sequence. The first three repeated data octets, which are actually sent, signal the beginning of a run-length encoded sequence. The next octet is always a repetition count which has a maximum value of 250. If the repeated sequence is only three octets in length, a repetition count of 0 is used. Four octets consisting of three repeated data octets and a count octet are thus used to compress any repeating sequence from 3 to 250 octets. Figure 4.55 illustrates the format of MNP Class 5 run-length encoding as well as a few examples of its operation.

MNP Class 5 run-length encoding can be considered as the first level of a two-level data compression scheme. The second level of compression is known as adaptive frequency encoding and is applied to the data stream after any repeated data octets are removed by the use of run-length encoding.

In adaptive frequency encoding a compression token is substituted in the data stream for the actually-occurring data octet in an attempt to transmit fewer than eight bits for each data octet. The token used changes with the frequency of

Format

data flow

| count | $O_3$ | $O_2$ | $O_1$ |

Examples

data string

------   IN    Compression   OUT
********        Routine

compressed data string

6---
8***
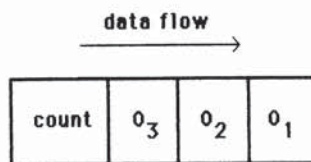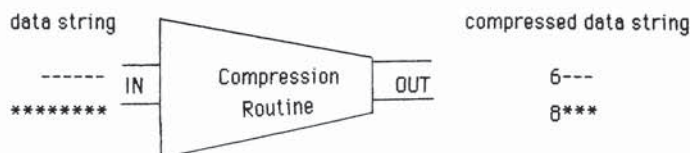
**Figure 4.55**   MNP class 5 run length encoding ($O_1 = O_2 = O_3$ = repeating octets)

**Table 4.26** MNP Class 5 octet/token mapping at compression initialization

| Data octet (decimal value) | Header (MSB → LSB) | Body (MSB → LSB) |
|---|---|---|
| 0 | 000 | 0 |
| 1 | 000 | 1 |
| 2 | 001 | 0 |
| 3 | 001 | 1 |
| 4 | 010 | 00 |
| 5 | 010 | 01 |
| 6 | 010 | 10 |
| 7 | 010 | 11 |
| 8 | 011 | 000 |
| 9 | 011 | 001 |
| 10 | 011 | 010 |
| 11 | 011 | 011 |
| 12 | 011 | 100 |
| 13 | 011 | 101 |
| 14 | 011 | 110 |
| 15 | 011 | 111 |
| 16 | 100 | 0000 |
| 17 | 100 | 0001 |
| 18 | 100 | 0010 |
| 19 | 100 | 0011 |
| 20 | 100 | 0100 |
| 21 | 100 | 0101 |
| 22 | 100 | 0110 |
| 23 | 100 | 0111 |
| 24 | 100 | 1000 |
| 25 | 100 | 1001 |
| 26 | 100 | 1010 |
| 27 | 100 | 1011 |
| 28 | 100 | 1100 |
| 29 | 100 | 1101 |
| 30 | 100 | 1110 |
| 31 | 100 | 1111 |
| 32 | 101 | 00000 |
| 33 | 101 | 00001 |
| 34 | 101 | 00010 |

[35–246-token header/body continue in same pattern]

| | | |
|---|---|---|
| 247 | 111 | 1110111 |
| 248 | 111 | 1111000 |
| 249 | 111 | 1111001 |
| 250 | 111 | 1111010 |
| 251 | 111 | 1111011 |
| 252 | 111 | 1111100 |
| 253 | 111 | 1111101 |
| 254 | 111 | 1111110 |
| 255 | 111 | 11111110 |

occurrence of the actual data octet so that shorter tokens are substituted for more frequently occurring data octets.

The compression token used to represent a data octet is composed of two parts: a fixed-length header and a variable-length body. The header is three bits long and, in general, indicates the length of the body portion of the token. There are three special cases, however. First, there are two tokens with a header of 0. In these two cases, the true length of the body is 1. Finally, when the header indicates a length of 7 and the body is seven 1-bits, then the actual length is 8.

At initiation of data compression, the relative frequency of occurrence of each data octet is 0. For purposes of octet/token mapping, however, data octet 0 (00000000 binary) is assumed to be the most frequently occurring octet and is represented by the first of the shortest tokens, data octet 1 (00000001 binary) is taken to be the next most frequently occurring and is represented by the next token, etc., to data octet 255 (11111111 binary), initially assumed to be the most infrequently occurring data octet. Data octet/token mapping at compression initialization is shown in Table 4.26

In order to encode a data octet, the token to which it is currently mapped is substituted for the actual data octet in the data stream. After this substitution, the frequency of occurrence of the current data octet is increased by one. If the frequency of this data octet is greater after incrementing than the frequency of the next most frequently occurring data octet, then the compression tokens of the current data octet and the next most frequently occurring data octet are exchanged. The frequency of the current data octet is then compared to the frequency of the data octet which is now the next most frequently occurring data octet. If the frequency of the current data octet is greater, then the compressed tokens are once again swapped. This cycle continues until no more swaps are needed at which time the mapping of data octets and compression tokens is correctly adapted based on the relative frequency of the data octets.

Once the data octet/compression token mappings are sorted by frequency, the frequency count of the current character is compared to the fixed limit value of 255 (decimal). If this limit has been reached, then the frequency of occurrence of each data octet is scaled downward by dividing each frequency by 2. Integer division is used, thus any remainder after division is discarded (e.g., $3/2 = 1$).

As previously noted, the repetition count, the fourth octet in a run-length encoded sequence of repeated data octets, is also mapped to a compression token. The token used is that which is mapped to the count of the most frequently occurring data octet. The token used for a count of 5 would thus be 01001. Note also, that this count octet does not increase the frequency of occurrence of the data octet to which the token is mapped. Further, the repetitions of the run-length encoded data octet represented by the count do not contribute to the frequency of occurrence of the repeated data octet.

## MNP Class 7 enhanced data compression

MNP Class 7 enhanced data compression builds upon the concept of combining run-length encoding with the use of an adaptive encoding table that contains a

single column representing each character ordered by their frequency of occurrence. Under MNP Class 7, run-length encoding is combined with the use of a first-order Markov model. This model is used to predict the probability of the occurrence of a character based upon the value of the previous character, with an adaptive table of 256 character columns used to represent the ordered frequency of occurrence of each 'succedding' character.

## Markov model

To encode a character the compressor selects a code which depends on the immediately preceding encoded character. The selected code is based on the frequency with which a character follows the previous character. For example, the probability of a 'U' following a 'Q' is very high and generally the 'U' will be encoded as 1 bit. Likewise, an 'H' following a 'C' has a different probability than an 'R' following a 'C', and will be coded according to its frequency of occurrence.

As noted in Table 4.27, the compressor keeps up to 256 coding tables, one for each possible 8-bit character (or pattern). To code a character for transmission, it uses the previous character to select the appropriate coding table. For example, when an 'A' is transmitted the model looks under the 'A' pointer for the next character, which is ordered according to its frequency of occurrence. If a 'C' is the next character, it is compressed based upon its location in the table. The model looks next under the 'C' pointer to find the following character, and so on. Each table contains the codes for characters following the previous character, and is organized according to the rules of Huffman coding.

**Table 4.27** MNP Class 7 enhanced data compression

| Pointers | A | B | C | D | E | | |
|---|---|---|---|---|---|---|---|
| Characters coded | T | L | H | O | D | •• | Up to |
| according to their | H | E | O | A | R | | maximum |
| frequency of following | C | U | R | E | S | | of 256 |
| the previous character, | M | . | | . | N | | characters |
| i.e., the pointer | B | . | | . | P | | each |

## Huffman coding

In Table 4.27, each column of characters under the pointer character is compressed according to the rules of Huffman coding. Huffman coding changes the number of bits representing a character when the character's frequency of occurrence changes sufficiently. Huffman can adapt to various alphabets (e.g., ASCII, EBCDIC, and all uppercase) and languages (natural language, compiler code, and spreadsheets) without being pre-informed of the data used.

Unlike Class 5, Huffman coding can represent a character with only one bit, if it occurs often enough. In general, if one character occurs twice as often as another, its code is half as long.

MNP Class 7 is adaptive, meaning that it changes the coding of the data when the frequency of character occurrence changes. The compressor starts off with no

assumptions about the data coding, and even learns that it is ASCII/English based on the data itself. The compression tables are empty at the start of each connection and are built as data is pased. When the data divides naturally into characters and the 'working set' of frequent characters is not too large (natural language is good here), then adaptation will create a coding structure which compresses the data well.

### Run-length encoding

Multiple consecutive copies of the same character (or 8-bit pattern) are compressed for transmission using run-length encoding. If the encoder has sent the same character three consecutive times, the encoder sends the count of the remaining identical consecutive characters as a single 4-bit nibble. For example, a series of 'A's would be sent as 'AAA' with the remaining number of 'A's sent as a 4-bit nibble, while, a series of five 'A's would be sent as three Huffman encodings of 'A' and a nibble of binary two.

### Decoding

To decode an encoded data stream, the receiving modem assumes it has an exact copy of the sending modem's compression data structures because:

(i) both modems reset their structures at the start of each connection;
(ii) communication has been error-free (MNP assures this);
(iii) decoding the data stream causes identical data structure updates to match the encoder's.

Since the receiving modem has the same compression tables as the sender, it knows which set of Huffman data decodes the incoming bit stream. The receiving modem compares the incoming bits against the Huffman codes in the table until there is a complete match. At this time, the character is delivered to the DTE and the table is updated.

## V.42 bis

V.42 bis data compression is a modified version of the Lempel-Ziv method of compression that was developed approximately 20 years prior to the promulgation of the recommendation. The V.42 bis compression method uses an algorithm in which a string of information received from an attached DTE is encoded as a variable length codeword. To facilitate the development of codewords strings are stored in dictionaries at the encoding and decoding device and are dynamically updated to reflect changes in the composition of data.

The key to the efficiency of the V.42 bis compression process is its dictionary which is dynamically built and modified. The dictionary can be considered to represent a set of trees in which each root corresponds to a character in the alphabet, as illustrated in Figure 4.56. Each tree represents a set of known strings beginning with one specific character, and each node or point in a tree represents
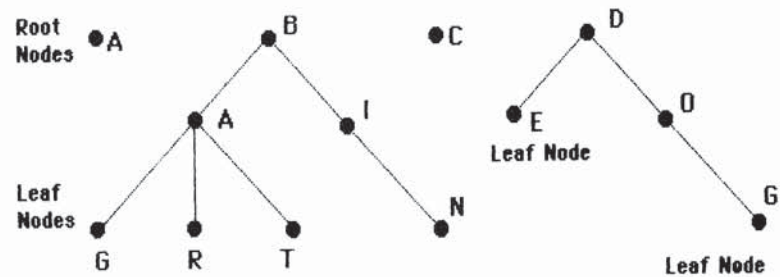
**Figure 4.56** V.42 tree-based dictionary representation

one set of strings. The trees in Figure 4.56 thus represent the strings A, B, BA, BAG, BAR, BAT, BI, BIN, C, D, DE, DO and DOG.

Each of the leaf nodes shown in Figure 4.56 represents a node that has no other dependent nodes, in effect, representing the last character in a string. Conversely, a node that has no parent is known as a root node and represents the first character in a string.

Initially, each tree in the dictionary consists of a root node, with a unique codeword assigned to each node. As data is received from an attached DTE a string matching process occurs in which a sequence of DTE originated characters is matched against the dictionary. This process begins with a single character. If the string matches a dictionary entry and the entry was not created since the last invocation of the string matching procedure, the next DTE originated character is read and appended to the string. This process is repeated under the previously described conditions until the maximum string length is reached or the string does not match a dictionary entry or matches an entry created since the last invocation of the string matching procedure. At that time the last character appended to the string is removed and represents an 'unmatched' character while the characters in the string are encoded as a codeword.

Under V.42 bis the maximum string length can range from 6 to 250 and is negotiated between modems. The number of codewords has a default value of 512, which is its minimum value. A maximum value is not, however, specified and any value above the default value can be negotiated between two modems.

During the compression process the dictionary is dynamically modified by the addition of new strings based upon the compostion of the data. New strings are formed by appending a single character that was not matched from a string matching operation to an existing string, which results in the addition of a new node to a tree. As a result of the replacement of strings by codewords V.42 bis data compression is approximately 20 to 30% more efficient than MNP Class 5 compression and probably 5 to 10% more efficient than MNP Class 7 enhanced data compression. While the increased efficiency of V.42 bis can be expected to result in many modem manufacturers providing this capability in future products, for many modem users the difference between V.42 bis and either MNP Class 5 or Class 7 operations may not be perceptible. This is because interactive access to mainframe computers normally involves queries based upon short strings of information that are not very susceptible to data compression. In comparison, users with extensive

file transfer requirements will find a more perceptible difference favouring V.42 bis. Thus, you should carefully consider their transmission requirements prior to purchasing a modem that peforms data compression as MNP Class 5 or Class 7 may be efficient as V.42 bis for certain applications.

## Compatibility issues

The adoption of the V.42 bis recommendation resulted in some vendors providing V.42 bis chip set upgrades for their products. Other vendors announced that all future products will incorporate V.42 bis support. Although V.42 bis is the recommended standard, unlike V.42 it does not specify MNP as an alternative method of data compression. Recognizing that over 15 million modems have been sold that support MNP Class 5 or Class 7 data compression, most modem manufacturers have incorporated both MNP Class 5 and V.42 bis compression into their products. Microcom offers both MNP Class 7 and V.42 bis support. Vendors other than Microcom may be limited to Class 5, because at the time this book was prepared, MNP Classes 1 through 4 were in the public domain and different licensing arrangements were applicable to the higher levels of the MNP protocol.

Modems that support both V.42 bis and MNP Class 5 data compression provide you with the door to the future as well as a gateway to the past. In the future you can expect that most, if not all, data compression-performing modems will be V.42 bis compatible. In the interim, MNP Class 5 compatibility provides modems with the ability to operate in a data-compressed mode with over 15 million currently installed modems.

## Throughput issues

No doubt many readers are familiar with the automotive section of the Saturday or Sunday newspaper in which the performance of a particular car is reviewed. Such reviews describe engine RPMs, torque, and other performance parameters that, for the most part, are irrelevant to the average driver. A similar case can be made for many reviews of data compression-performing modems in trade publications: the author discusses how a file was uploaded or downloaded in so many seconds due to the high operating rate and compression efficiency of the modem. What many modem reviews fail to mention is that it is extremely rare for you to obtain that level of performance for any sustained period of time during a typical communications session. In fact, the use of the compression feature of a modem may actually result in an increase in the time required to transfer certain types of files.

Although a 2:1 modem compression ratio means that you can expect to achieve twice the modem's operating rate for file transfer operations, it is assumed that you are transferring a standard ASCII text file. In actuality, most information utilities and bulletin board systems store files in a compressed format, using such programs as ARC, ARJ, PAK, and PKZIP to economize on file storage space. Attempting to use modem compression to transfer a previously compressed file can actually result in an expansion of the file during data transfer between modems if you are using

MNP5, or a marginal improvement in data transfer time if you are using the V.42 bis data compression method. Thus, a good rule of thumb is to transfer compressed files without using MNP5 data compression if that is the only compression method supported by your modem.

## Negotiation problems

As previously indicated in this section, both error detection and correction and data compression are negotiated by the flow of data between modems once the basic modulation handshaking process is completed. This data flow occurs through the transmission of frames between modems. These frames describe the settings and capabilities of each modem. Although MNP, V.42, and V.42 bis all explicitly define how data frames are encoded and exchanged, significant problems can occur if one modem that has negotiation features enabled communicates with another modem that either does not have such capabilities or has those capabilities disabled. When either situation occurs, the negotiation frame will be passed onward by the modem that does not have enabled negotiation capabilities. If the modem is connected to an encryption device, data PBX, or another device that is data sensitive, the negotiation frame can cause havoc.

Figure 4.57 illustrates one example of *modem negotiation havoc*. In this example a user with a notebook containing a built-in datacryptor uses the switched telephone network to communicate in a secure manner with a mainframe computer. If the modem attached to the notebook computer has its negotiation capabilities enabled,
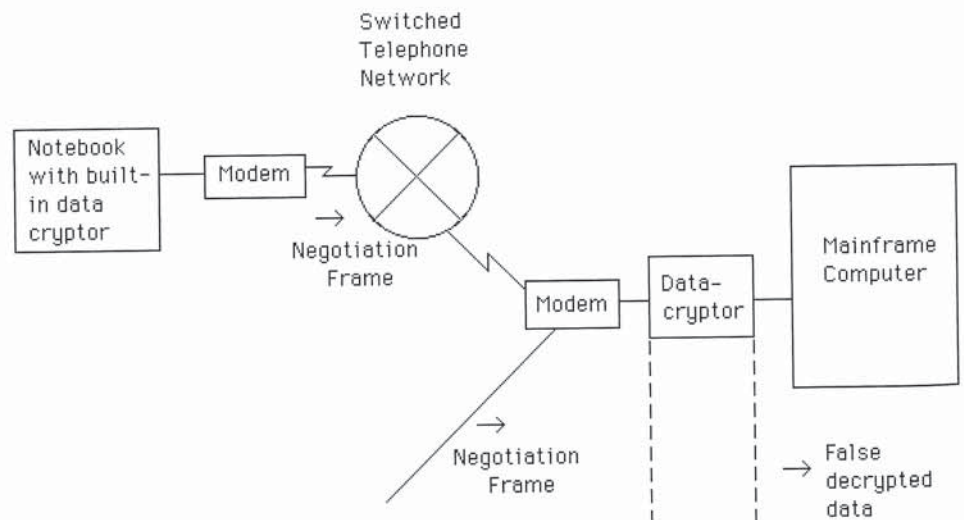


**Figure 4.57** Modem negotiation havoc. When one modem transmits negotiation frames and the other either has its negotiation capability disabled or lacks a negotiation capability, the frames are passed through the modem as data. In this network configuration, the datacryptor performs its intended operation, resulting in false decrypted data sent to the mainframe, which results in a loss of synchronization between datacryptors

upon establishment of a carrier signal it will initiate handshaking by transmitting a negotiation frame. If the distant modem either lacks a negotiation capability or has that feature disabled, the negotiation frame passes through the modem as it is assumed to represent demodulated data. Unfortunately, the datacryptor also assumes it is demodulated data and initiates its decryption operation. Not only does this result in false decrypted data being passed to the mainframe but, in addition, it results in a loss of synchronization between the pair of datacryptors. Thus, when the notebook user attempts to transmit a request to the mainframe, data is encoded based upon one position in a pseudo-random key sequence generated by the datacryptor built into the notebook, and decoded at the mainframe by the datacryptor using a different position in a pseudo-random key sequence since the negotiation frame preceded the data. Needless to say, the mainframe receives what appears to be 'gobbledegook'.

Another common example of modem negotiation havoc can occur in a data PBX environment. When one modem transmits negotiation frames and the other either has its negotiation capability disabled or lacks a negotiation capability, the frames are passed through the modem as data. In this network configuration the datacryptor performs its intended operation, resulting in false decrypted data being sent to the mainframe which results in a loss of synchronization between datacryptors. If a data PBX is behind a modem, the negotiation frame could be misinterpreted as a routing code, and could cause a misrouting before a user could enter a correct routing code. To correct such situations, you can enable error detection and correction and data compression features on modems at the other end of the communications link, if those features were previously disabled, or replace modems that don't support those features with modems that do. A third option is to disable those features on the modem transmitting the negotiation frame. However, doing so eliminates the possibility of using those features.

## Simultaneous voice and data operations

The growth in 'telecommuting' has resulted in many people working from their homes, accessing LAN servers and corporate mainframes to perform work that might otherwise require a long commute into the office. Accompanying the growth in telecommuting is an increase in the requirement to communicate verbally with other telecommuters or with staff at the office while maintaining a data transmission session. Although the Integrated Services Digital Network (ISDN) provides the ability to maintain two separate conversations on one pair of telephone wires, enabling a voice and a data version to coexist simultaneously, the technology is relatively expensive in comparison to the cost of using the analog PSTN and is far from ubiquitous. Recognizing those limitations, as well as the requirements of telecommuters to converse simultaneously with a distant party while transmitting data, resulted in the development of simultaneous voice and data (SVD) capability into certain modem products.

Currently two methods are used to provide a simultaneous voice and data capability. One method, referred to as *digital simultaneous voice and data* (DSVD), uses speech compression technology to digitize voice into a low bit rate which

enables users to carry on a normal conversation over the PSTN while transmitting data at rates up to 33.6 kbps. The second method results in the transmission of voice in its native, analog format, while reducing the data rate to provide separate channels for voice and control information. This method is commonly referred to as analog simultaneous voice and data (ASVD). Although ASVD can be more aesthetically pleasing to a listener's ear in certain situations, most implementations of a SVD capability work digitally to obtain a higher data transfer operating rate capability. Two competitive and incompatible techniques for obtaining a simultaneous voice and data transmission capability were being offered by Multi-Tech Systems and Rockwell Semiconductor Systems when this book was published.

Multi-Tech Systems markets a series of DSVD modems using codecs (coders-decoders) to digitize the full spectrum of audio bandwidth from 0 to 3300 Hz.

Using what is referred to as the Multi-Tech Supervisory Protocol (MSP) for communications between modems, voice, data, and supervisory information are separately packetized for transmission. All three types of packets are multiplexed onto a single stream of data which enables a common circuit to support voice and data simultaneously; however, to eliminate potential voice distortion voice packets receive priority over data packets. Figure 4.58 illustrates the MSP packet structure. Note that each packet uses one 8-bit SYNC byte header for synchronization.

The ID/LI Information byte includes an ID (bit number 7) and a Packet Length Indicator (LI) field which ranges in value from 01h to 07Fh for data and qualified packets. Voice packets are preset to 24 or 39 bytes in length based upon the data operating rate. Thus, the ID/LI Information byte is set to a value of 00h in voice packets.

A. Data Packet

| 01h SYNC | ID LI | DATA Byte 1 | . . . . . . . . . . | DATA BYTEX |
|----------|-------|-------------|---------------------|------------|

B. Voice Packet

| 01h SYNC | 00h ID | DATA Byte 1 | . . . . . . . . . . | BYTE 24=9.6K 39=16K |
|----------|--------|-------------|---------------------|---------------------|

C. Qualified Packet

| 01h | ID LI | QUAL BYTE | DATA Byte 1 | DATA BYTEX |
|-----|-------|-----------|-------------|------------|
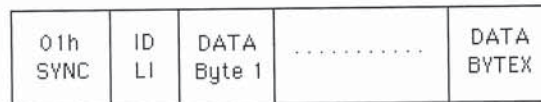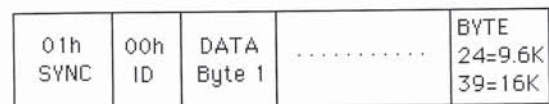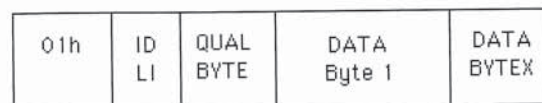
**Figure 4.58**  Multi-Tech Supervisory Prtotocol packet structure

The Qualifier (QUAL) byte, which is present in Qualified packets, identifies the nature of the command that follows that byte. Those commands fall into three broad categories—'immediate action' commands, 'engage stream mode for' commands, and 'interpret the following as' commands. Examples of 'immediate action' commands include ACK, NAK, XON, XOFF, and BREAK. Examples of 'engage stream mode for' commands include COMMANDS, VOICE and DATA, while 'interpret the following as' commands include commands, responses and status. Thus, the MSP functions in a manner similar to the Higher-Level Data Link Control (HDLC) protocol in that it supports command/response transmission sequences.

Although Multi-Tech's MSP provides simultaneous voice with data at an operating rate up to 16 kbps, the protocol does not attempt to detect and take advantage of periods of voice silence. A competitive simultaneous voice and data product from Rockwell known as DigiTalk and incorporated into that vendor's RC288ACi/SVD chip set both digitizes voice into a low bit rate and detects periods of silence, enabling modems using the chip set to shift to higher data operating rates during periods of voice silence. This technique permits a V.34 modem using a DigiTalk chip set to obtain a non-compressed data transfer rate of 28.8 kbps during periods of voice silence.

## Synchronous dialing language

In comparison to the Hayes command set that is a _de facto_ standard for asynchronous modem operations there are several methods used to support synchronous modem operations. The earliest method of automating synchronous dialing was accomplished by the use of an automatic calling unit. In this method of establishing calls over the PSTN a computer or another DTE device is connected to an automatic calling unit via an RS-366 (parallel) or an RS-232 (serial) interface. The DTE controls the calling unit, which dials the telephone number and upon completion passes the call to the modem. Figure 4.59 illustrates this procedure.

Other methods that have been developed to obtain synchronous dialing capability include the incorporation of an RS-366 interface into some modems, the use of an integrated external keypad through which the operator stores dialing sequences which the modem dials when the attached DTE raises the Data Terminal Ready signal and the use of a synchronous dialing language which
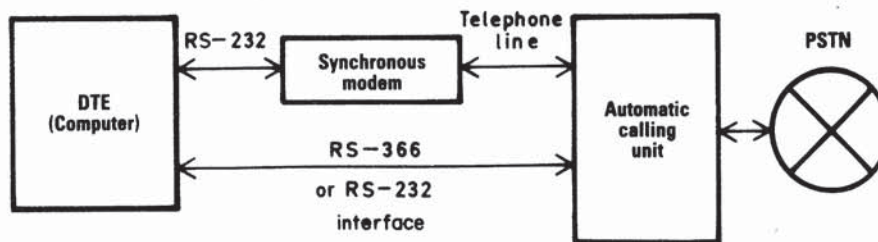
**Figure 4.59** Using an automatic calling unit. Automatic calling units support both RS-232 (serial) and RS-366 (parallel) interfaces

combines dial control and data over a common RS-232 interface. Concerning the latter, there are currently several methods competing to become *de facto* industry standards.

Hayes Microcomputer Products' proposed standard is the Hayes Synchronous Interface (HSI). This interface consists of a hardware independent connection between a user's application program and a synchronous communications driver which directly controls the hardware. Under the HSI method, dialing commands to a synchronous modem are issued asynchronously, with the modem directed to change into a synchronous mode of operation once a conncetion with a distant device is made. Obviously, this method is only applicable for use with modems that can work both synchronously and asynchronously; however, it does permit the ubiquitous Hayes AT command set to be used.

A second method being used to perform automatic dialing operations with synchronous modems is based upon the use of the Racal-Vadic Synchronous Auto-Dial Language (SADL). Under SADL special modem controlling messages in both bisynchronous and SDLC protocols are predefined. This enables software developers to modify application programs to support the use of SADL compatible modems for dialing on the PSTN.

The third method being used for synchronous dialing is similar to the previously described SADL technique. This method is called 'SyncUp', which is sponsored by Universal Data Systems and is restricted to sending dialing commands under the bisynchronous protocol.

## 4.4 MULTIPORT MODEMS

The integration of modems and limited-function multiplexers into a device known as a multiport modem offers significant benefits to data communications users who require the multiplexing of only a few channels of data. Users who desired to multiplex a few high-speed data channels prior to the introduction of multiport modems were required to obtain both multiplexers and modems as individual units which were then connected to each other to provide the multiplexer and data transmission requirements of the user. Since multiplexers are normally designed to support both asynchronous and synchronous data channels, the cost of the extra circuitry and the additional equipment capacity was an excess burden for many user applications.

The recognition by users and vendors that a more cost-effective, less wasteful method of multiplexing and transmitting a small number of synchronous channels for particular applications led to the development of multiport modems. By the combination of the functions of a time division multiplexer (TDM) with the functions of a synchronous modem, substantial economies over the past data transmission methods can be achieved for certain applications.

### Operation

A multiport modem is basically a high-speed synchronous modem with a built-in TDM that uses the modem's clock for data synchronization, rather than requiring

one of its own, as would be necessary when separate modems and multiplexers are combined. In contrast with most traditional TDMs, a multiport modem multiplexes only synchronous data streams, instead of both synchronous and asynchronous data streams. An advantage of the built-in, limited-function multiplexer is that it is less complex and expensive, containing only the logic necessary to combine into one data stream information transmitted from as few as two synchronous data channels rather than the minimum capacity of four or eight channels associated with most separate multiplexers. The data channels in a multiport modem normally comprise a number of 2400 bps data streams, with the number of channels available being a function of the composite transmission speed.

## Selection criteria

When investigating the potential use of multiport modems for a particular application, you should determine the speed combinations and the number of selectable channels available, as well as the ability to control the carrier function (mode of operation) independently for each of the channels. For example, one 9600 bps multiport modem can have as many as six different modes of operation; however, only one mode can be functioning at any given time. As illustrated in Figure 4.60, operating speeds can range in combination from a single channel at 9600 bps through four 2400 bps channels.
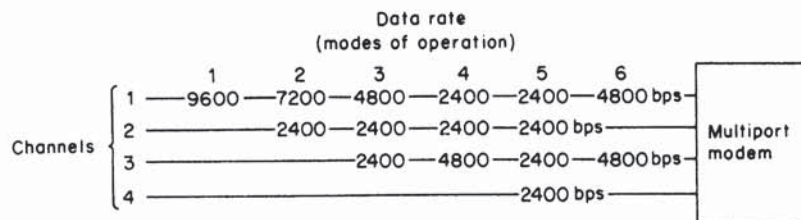


**Figure 4.60** 9600 multiport modem schematic utilization diagram. Multiport modem with six modes of operation is schematized here to show all possible data rate combinations for networking flexibility

## Application example

A typical application of a 28 800 bps multiport modem is illustrated in Figure 4.61. This example shows a pair of four-channel 28 800 bps multiport modems servicing two CRT terminal devices, a synchronous printer operating at up to 1500 characters per second or 12 000 bps, and three low-speed synchronous terminals connected by a traditional TDM. The output of the 3-channel TDM is a 7200 bps synchronous data stream, which is in turn multiplexed by the multiport modem. Here the multiport modem's multiplexer combines the three asynchronous
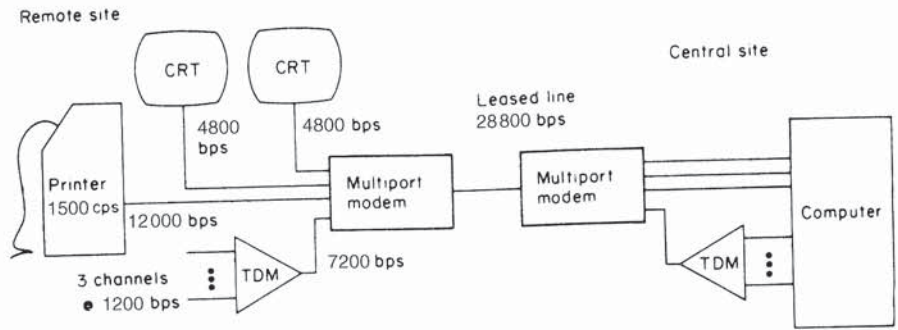
**Figure 4.61** Multiport modem application example. A pair of four-channel multiport modems services two CRTs, a 1500 cps printer, and three low speed terminal devices over a single transmission line

multiplexed 1200 bps channels with the three synchronous unmultiplexed channels into a single multiplexed synchronous data stream. At the central site where the computer is located, the multiport modem at that end splits the 28 800 bps stream into four data streams; one data stream is then channeled through another 3-channel, traditional TDM, whose output data streams in turn are connected to the computer. The 3-channel TDM takes the 7200 bps synchronous data stream from the multiport modem and demultiplexes it into three 1200 bps asynchronous data streams, which are passed to the appropriate computer ports. The remaining data streams produced by the demultiplexer in the multiport modem are connected to three additional computer ports. As this example demonstrates, the high-speed multiport modem's utilization in conjunction with other communications components permits a wide degree of flexibility in the design of a data communications network.

A few examples of multiport modem and channel combinations available are listed in Table 4.28. It should be noted that not all multiport modem channel combinations listed in Table 4.28 may be available for a particular vendor's modem that operates at the indicated aggregate throughput, since some vendors offer only a 4-port multiplexer with their modem, while other vendors may offer an additional number of ports.

Although most manufacturers of multiport modems produce equipment that appears to be functionally equivalent, you should exercise care in selecting equipment because of the differences that exist between modems but are hard to ascertain from vendor literature. Although the modes of operation available for utilization are quite similar regardless of manufacturer, the number of ports or data channels supported by a multiplexer can differ significantly. Another variance concerns the use of an independently controlled carrier signal in some multiport modems. By using multiport modems that permit an independently controlled carrier signal for each channel, you can combine several polled circuits and further reduce leased line charges.

**Table 4.28** Examples of multiport modem channel combinations. The wide degree of flexibility that can be provided by multiport modems in a network configuration is a function of several factors: throughput, available modes and channels, and data rates

| Modem aggregate throughput | Operating mode | Multiport speed combinations | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| | | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
| 19 200 | 1 | 19200 | | | | | | | |
| | 2 | 16800 | 2400 | | | | | | |
| | 3 | 14400 | 4800 | | | | | | |
| | 4 | 14400 | 2400 | 2400 | | | | | |
| | 5 | 12000 | 7200 | | | | | | |
| | 6 | 12000 | 4800 | 2400 | | | | | |
| | 7 | 12000 | 2400 | 2400 | 2400 | | | | |
| | 8 | 9600 | 9600 | | | | | | |
| | 9 | 9600 | 4800 | 4800 | | | | | |
| | 10 | 9600 | 4800 | 2400 | 2400 | | | | |
| | 11 | 7200 | 7200 | 4800 | | | | | |
| | 12 | 7200 | 4800 | 2400 | 2400 | 2400 | | | |
| | 13 | 7200 | 2400 | 2400 | 2400 | 2400 | 2400 | 2400 | |
| | 14 | 4800 | 4800 | 4800 | 4800 | | | | |
| | 15 | 4800 | 4800 | 4800 | 2400 | 2400 | | | |
| | 16 | 4800 | 4800 | 2400 | 2400 | 2400 | 2400 | | |
| | 17 | 4800 | 2400 | 2400 | 2400 | 2400 | 2400 | 2400 | |
| | 18 | 2400 | 2400 | 2400 | 2400 | 2400 | 2400 | 2400 | 2400 |
| 14 400 | 1 | 14400 | | | | | | | |
| | 2 | 12000 | 2400 | | | | | | |
| | 3 | 9600 | 4800 | | | | | | |
| | 4 | 9600 | 2400 | 2400 | | | | | |
| | 5 | 7200 | 7200 | | | | | | |
| | 6 | 7200 | 4800 | 2400 | | | | | |
| | 7 | 7200 | 2400 | 2400 | 2400 | | | | |
| | 8 | 4800 | 4800 | 4800 | | | | | |
| | 9 | 4800 | 4800 | 2400 | 2400 | | | | |
| | 10 | 2400 | 2400 | 2400 | 2400 | 2400 | 2400 | | |
| 9600 | 1 | 9600 | | | | | | | |
| | 2 | 7200 | 2400 | | | | | | |
| | 3 | 4800 | 2400 | 2400 | | | | | |
| | 4 | 2400 | 2400 | 4800 | | | | | |
| | 5 | 2400 | 2400 | 2400 | 2400 | | | | |
| | 6 | 4800 | 4800 | | | | | | |
| 7200 | 1 | 7200 | | | | | | | |
| | 2 | 4800 | 2400 | | | | | | |
| | 3 | 2400 | 2400 | 2400 | | | | | |
| 4800 | 1 | 4800 | | | | | | | |
| | 2 | 2400 | 2400 | | | | | | |

## Standard and optional features

A range of standard and optional data communications features are available for users of multiport modems, including almost all of the features available in regular non-multiplexing modems; also available are unique multiport modem features such as multiport configuration selection, individual port testing, individual port display, and a data communications equipment (DCE) interface.

The multiport selector feature permits you to alter the multiport configuration simply by throwing a switch into a new position. This feature can be especially useful for an installation such as the one shown in Figure 4.62 (top) where communications requirements differ between shifts. For example, in the figure, during daytime operations three low-speed asynchronous, 4800 bps, terminal devices with a composite speed of 19 200 bps are serviced at this installation by one channel of
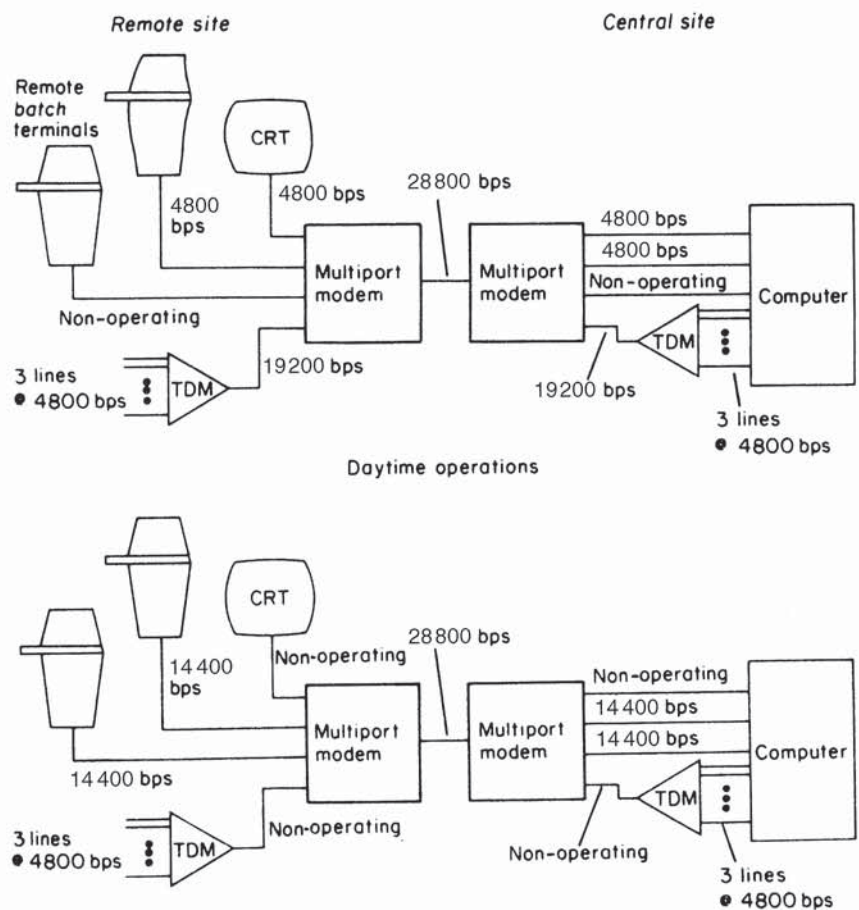
**Figure 4.62** Using multiport selector switches. Dayy (top) and night (bottom) configurations for networks with mutiport modems can be varied according to the requirements of different operations

the multiport modem. One remote batch terminal and a CRT are serviced by two additional channels, each of which operates at 4800 bps. Because of daytime load requirements, the second batch terminal cannot be operated since the modem's maximum aggregate speed of 28 800 bps has been reached. On the assumption that the installation does not require the servicing of interactive users at night, one possible reconfiguration is shown in Figure 4.62 (bottom). The multiport selector permits both remote batch terminals to be serviced until the start of the next business day by two 14 400 bps channels while everything else in this network is shut down.

Numerous multiport modems contain a built-in test pattern generator and an error detector which permit users of such modems to determine if the device is faulty without the need of an external bit error rate tester. The use of this feature normally permits the individual ports of the modem to be tested.

Another option offered by some multiport modem manufacturers is a data communications equipment (DCE) interface. This option can be used to integrate remotely located terminals into a multiport modem network. Whereas the standard data terminal equipment (DTE) interface may require data sources to be collocated and within a 50-foot radius of the multiport modem, the DCE interface permits one or more data sources to be remotely located from the multiport modem. Installation of a multiport modem with a DCE option on one port permits that modem's port to be interfaced with another modem. This low-speed conventional modem can be used to provide a new link between the multiport modem's location and terminals located at different sites. As shown in Figure 4.63, the installation of a multiport modem with a DCE option on port 3 permits the port on that modem to be interfaced with another modem. This new modem can then be used to provide a new link between the multiport modem at location 1 and an additional remote batch terminal which is located at a second site.
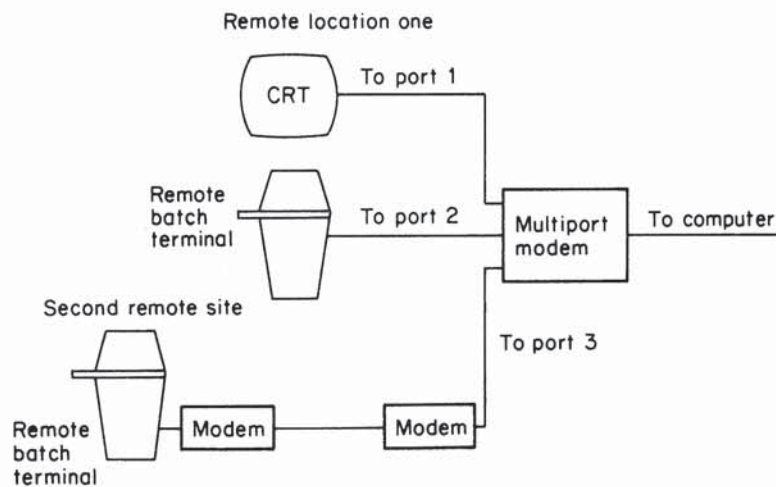
**Figure 4.63** Multiport modem data communications equipment option. Using a data communications equipment (DCE) interface on port 3 of the multiport modem permits a second remote site to share the communications line from the first site to the computer

## 4.5 MULTIPOINT MODEMS

To alleviate substantial confusion, it should be noted that a multiport modem contains a built-in multiplexer which enables two or more separate data streams to be combined for transmission over a single circuit. In contrast, a multipoint modem is basically a modem designed to achieve fast polling acquisition times on multipoint lines.

Multipoint lines, or multidrop lines as they are also referred to, are usually installed for applications that require interactive terminal access from a number of geographically dispersed locations into a central computer facility.

An airline reservation system in which dispersed terminals randomly access the computer's database to determine flight information and seat availability is a representative application that could use multipoint lines. Another example would be an inventory control system where terminals are located at many warehouses and are used to report shippings and arrivals so that company inventories are continuously updated. For either application, the key item of interest is that each terminal only uses a small fraction of the total time available to all terminals connected to the line to complete a transaction, and the terminal is addressable and can recognize messages for which it is a recipient.

Although most terminals connected to multipoint circuits contain a buffer, this buffer primarily serves to enhance data throughput and is not a necessity. For example, consider the situation where an operator is entering data on the screen of a CRT through an attached keyboard. If the CRT does not have a buffer, the time it takes to transmit the data depends upon the operator's speed in typing it. During that time communications with all other terminals on the line are suspended. If the terminal has a buffer, the transmission speed from the terminal to the computer can be at a much higher rate than the operator's typing rate. Thus, once the operator has filled the CRT screen with data, the depression of a transmit key will permit the computer to select and receive the data from the terminal at a higher transfer speed and in a shorter time interval.

During the time the operator is entering data the computer is free to select other terminals. The wait time per terminal on the common line is therefore reduced. Since a slow operator on a multidrop line without buffered terminals could obtain an unjustified proportion of the total transmission time, some multidrop systems
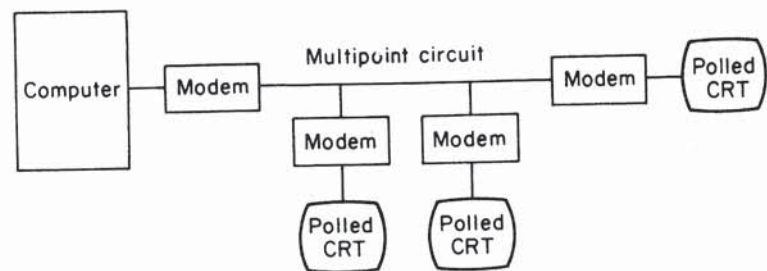


**Figure 4.64** Typical multipoint circuit. On a multipoint circuit many polled terminals share the use of common communications facilities

incorporating unbuffered terminals have a built-in time-out feature. This time-out feature permits another operator to gain control of the line if the first operator pauses for a time greater than the time-out feature permits. When either type of terminal is used in the previously described working environment, then many terminals can share the same communications circuit on an interleaved basis. The polling and selecting protocol used will make it appear to each terminal operator as if a private connection existed for his or her exclusive use for transmitting and receiving data from the computer. A typical multipoint circuit used to connect three terminals is illustrated in Figure 4.64.

## Factors affecting multipoint circuits

When the applicability of a multipoint circuit is being investigated, several parameters warrant careful investigation; two such parameters are the response time and the transaction rate of the terminals.

### Response time

From a broad viewpoint, response time is the time interval from when an operator presses a transmit key at the terminal until the first character of the response appears back at that terminal. This response time consists of the many delay times associated with the components on the circuit, the time required for the message to travel down the circuit to the computer and back to the terminal, as well as the processing time required by the computer.

### Transaction rate

The transaction rate is a term used to denote the volume of inquiries and responses that must be carried by the circuit during a specific period of time. This rate is normally expressed in terms of a daily average and as a peak for a specific period of the day.

Additional factors that affect the data transfer rates on multipoint circuits include the line protocol used and its efficiency, the transmission rate of the modems, and the turnaround time of the line. While there are many factors that contribute to multipoint line efficiency utilization, the focus of this section will be on multipoint modem characteristics which should be investigated to obtain a more efficient transmission process on such circuits.

### Delay factors

When a terminal connected to a computer via a multipoint circuit is polled, several factors, which by themselves may appear insignificant, accumulate to degrade response time.

In the transmission of a message or poll from the computer to a terminal on the multipoint line, the first delay encountered is caused by the modem's internal delay time ($D_m$). This is the delay that would be seen if the time between the first bit entering the modem and the first modulated tone put on the circuit were measured. For a poll transmitted from the computer to a terminal or a response transmitted from the terminal to the computer, total modem delay time is equal to two times the modem's internal delay time ($2D_m$), since the poll or message is transmitted through two modems. In this case, the internal delay time of the modem is equivalent to the delay one would measure if the modems were placed back to back and the time beween the first bit entering the first modem and the first bit demodulated by the second modem were compared. This is shown in Figure 4.65. Depending upon the type of modem used, this internal delay time can vary from a few milliseconds to 20 ms or more.



**Figure 4.65** Internal modem delay time. Time difference from first bit transmitted (1) to first bit received (2) is denoted as total internal modem delay time ($2D_m$)

The second delay time on a multipoint circuit is a function of the distance between the computer and the terminal with which it is communicating. This delay time is called transmission or propagation delay ($D_p$) and represents the time it takes for the signal to propagate down the line to the receiving location. Although this delay time is insignificant for short transmission distances, coast-to-coast transmission can result in a transmission delay of approximately 30 ms on a terrestrial circuit or 250 ms if transmission occurs via a satellite.

When a response to a transmitted poll or message is returned from the remote terminal, several additional delay factors are encountered. First, the terminal itself causes a delay, since some time is required for the terminal to recognize the poll and initiate a response. The terminal delay ($D_t$) is usually a few milliseconds, but can be considerably longer, depending upon the design of the terminal and the software protocol used.

Since every modem on the multipoint line shares the use of this common circuit, only one modem at any point in time may have its carrier on. The carrier of each modem is turned on in response to the connected terminal or computer raising its request to send (RTS) signal, which indicates that a message is to be transmitted. The modem at the opposite end of the transmission path will then require some time interval to recognize this signal and adjust its internal timing. If the modems have automatic and adaptive equalization, additional time is spent adapting the modem to the incoming signal's characteristics.

Once these functions are completed, the modem located at the opposite end of the transmission path will raise its clear to send (CTS) signal which is required in response to the RTS signal if a return message is to be transmitted. This delay time

is referred to as the request to send/clear to send delay time (RTS/CTS) and is usually denoted as $D_{R/C}$. Since the RTS/CTS delay time occurs twice in responding to a poll, the total RTS/CTS turnaround delay is $2D_{R/C}$. Disregarding terminal, propagation, and processing delays, the total turnaround delay attributable to the modems is $M_D$, where

$$M_D = 2(D_{R/C} + D_m)$$

Although the RTS/CTS delay varies by the type of modem used, delays can range from about 5 to 100 ms or more and normally cause most of the line's turnaround delay.

When multipoint networks were initially implemented, modem operating rates were at speeds of 2400 bps or lower. These modems were manually equalized and were adjusted and set at installation time. For this category of modems, the RTS/CTS delay ranges from about 10 to 20 ms, depending upon manufacturer. Owing to the advancement in modem transmission rates as well as new applications which require higher traffic rates and shorter response times, multipoint networks with 4800 bps modems were implemented. At this data rate, two types of modem equalization are used, the manual type previously discussed and that which is automatically equalized. In comparison to the static nature of manual equalization, automatic equalization permits the modem to continuously monitor and compensate for changing line conditions. An initial period of time is, however, required for the modem to 'train' on the signal each time the transmission direction reverses. For manually equalized modems, the RTS/CTS delay time is normally between 10 and 20 ms, since they are static in nature. For automatically equalized modems, the training time necessary for the modem to adapt to the incoming signal adds significantly to the RTS/CTS delay time, with such delays increased to 50 ms or more.

## Throughput problems

To illustrate the effect of the increased delay on data throughput, let us examine the transmission of a 50-character data block. If each character consists of eight data bits, then the data block would contain 400 bits of information. If 2400 bps modems are used for transmission, the actual time required to transmit the block would be 400/2400 or 166 ms.

Let us assume that the RTS/CTS delay time of the 2400 bps modems was 10 ms and their internal delay time was 5 ms. The total delay time associated with the transmission and acknowledgement of the data block would then be 2 X (10 + 5) or 30 ms. Since the time to transmit the data block is 166 ms and the delay time associated with the block is 30 ms, the overhead attributable to the delay is 30/(166 +30) or 15.3%. Given a modem operating rate of 2400 bps, the effective data transfer rate becomes:

$$2400 - (2400 \times 0.153) = 2032 \text{ bps}$$

Now let us examine the effect of doubling the modems' operating rate to 4800 bps. At 4800 bps, modems employing automatic equalization originally had an

RTS/CTS delay time of 50 ms. Assuming the internal delay time remained constant at 5 ms, the total delay time to transmit the data block and receive an acknowledgement would become

$$2 \times (50 + 5) = 110 \text{ ms}$$

When the modems operate at 4800 bps, the time to transmit a 400-bit block is 400/4800 or 83.3 ms. Then the overhead attributable to the delay times is 110/ (83.3 + 110) or 56.9%. Given the modem operating rate of 4800 bps, the effective data transfer rate becomes

$$4800 - (4800 \times 0.569) = 2069 \text{ bps}$$

Based upon the preceding analysis, it is obvious that the doubling of a multipoint modem's operating rate only resulted in an insignificant increase in the effective data transfer rate. If this situation were allowed to persist, most data communications users would never benefit from upgrading the operating rate of modems used on multipoint circuits. Fortunately, modem developments resulted in an increase in the effective data transfer rate that has a high correspondence to the increase in a modem's operating rate.

## Multipoint modem developments

Until the early 1970s, users were forced to trade off the benefits derived from automatic equalization with the longer RTS/CTS delay time that was obtainable through the use of a modem equipped with this feature. Fortunately, many modem manufacturers incorporated techniques that reduce the RTS/CTS delay time while permitting automatic equalization.

One manufacturer implemented a so-called 'gearshift' technique where data transmission begins at a rate of 2400 bps, using a modulation technique that does not require extensive equalization. This reduces the RTS/CTS delay to a level of about 9 ms, which is the delay time normally associated with manually equalized modems transmitting at 2400 bps. Next, as transmission proceeds at 2400 bps, the receiving modem automatically equalizes on the incoming signal. After an initial transmission of 64 bits of data is received at the 2400 bps data rate, the training cycle is completed and both the sending and receiving modems 'gearshift' up to the faster 4800 bps data rate to continue transmission. While this technique will reduce the RTS/CTS delay time, the actual number of data bits transmitted during an interval of time will depend upon the size of the message transmitted. This is because the first 64 bits of each message are transmitted at 2400 bps prior to the modem gearshifting to the 4800 bps data rate.

Another technique used to increase the number of bits of information transmitted was obtained by incorporating a microprocessor into a modem. This microprocessor is used to perform equalization and provides a very fast polling feature which increases the data traffic transmitted during a period of time when compared to the standard Bell System equivalent 4800 bps modems or the gearshift-type modem. This comparison is illustrated in Table 4.29.

**Table 4.29** Comparison of data traffic transmitted by 4800 bps multipoint modems

| Time (ms) | Data bits | | |
|---|---|---|---|
| | Modem with microprocessor | Modem using 'gearshift' technique | Bell System equivalent |
| 9 | 0 | 0 | 0 |
| 20 | 24 | 24 | 0 |
| 26 | 58 | 41 | 0 |
| 36 | 106 | 65 | 48 |
| 50 | 173 | 133 | 115 |

Although the use of multipoint modems has significantly decreased since their peak period of usage during the mid-1980s, they can still be used effectively and economically to interconnect locations with relatively low volume transmission requirements. However, if you anticipate using multipoint modems you should select products with a minimal RTS/CTS delay to obtain a higher effective data transfer capability.

### Remote multipoint testing

Since it is much more difficult to ascertain the cause of problems on a multipoint line than on a point-to-point circuit, modem manufacturers developed several testing features that are either incorporated into multipoint modems or offered as an option for end-user selection. One of the more common testing features offered with multipoint modems is remote testing. This feature results in each remote multipoint modem containing a unique address, permitting a central site modem to send an address code and command signal to the remote modems on the multipoint circuit, which will then place each remote site modem into an analog or digital loop-back mode of operation. By incorporating an address, each remote modem can then be tested from the remote site.

## 4.6 SECURITY MODEMS

The goal in the development of the security modem is to identify authorized users based upon the telephone number from which they originate calls on the PSTN. To accomplish this goal the security modem is designed to receive calls originated on the PSTN, interrogate the user's identity based upon a code or password, disconnect the user and then initiate a call on the PSTN to a predefined telephone number stored in memory in the modem. Since the security modem initiates a callback to the user attempting access, another name commonly associated with this device is a callback modem.

### Operation

A callback or security modem contains a battery powered buffer area into which the network administrator enters an access code and telephone number for users

authorized to dial a PSTN telephone number that is connected to the modem. Depending upon the security modem used, between four and twelve characters are commonly available for the authorization code.

Once a network user receives the authorization code and dials the number assigned to the security modem, that device first prompts them to enter their code. Assuming the code is entered correctly, the modem then disconnects the user from the line and originates a call to the telephone number associated with the access code.

Since the security modem simply verifies the telephone number of the originator of the call it should not be confused with data encryption devices that scramble transmission based upon a predefined algorithm. Thus, the protection of the security modem is relegated to verifying that the data call originator has an authorized access code and is located at an authorized telephone number.

## Memory capacity and device access

The memory capacity and network administrator access to a security modem differ considerably between devices. Most security modems have a buffer area that only permits the storage of three to five access codes and associated telephone numbers. A few security modems have the capacity to store up to approximately 20 access codes and associated telephone numbers, representing the maximum capacity of devices in this equipment category. Concerning network administrator access, most modems require the network administrator to first cable a terminal to the device to enter the access codes and telephone numbers assigned to each access code. Then, once this is accomplished, the security modem can be connected to the computer or multiplexer port it is designed to service. This type of modem is difficult to modify, since it requires the network administrator to place it out of service and recable a terminal to the device to alter previously entered data. Then, after modifications are made to the access codes and telephone numbers to add new users or delete previously authorized users, the network administrator can disconnect the terminal and recable the security modem to the device it services.

A second method of updating access codes and associated telephone numbers is available on other security modems. This method requires the system administrator to first enter a master access code when they initially attach a terminal to the device. Thereafter, the network administrator can use any terminal attached to a compatible modem and dial the security modem via the PSTN. Once the master access code is entered, the network administrator can remotely enter or modify access codes and their associated telephone numbers.

## Device limitations

Since it is a natural tendency of many persons to post access codes on terminals, the use of a security modem is only as good as the policy and enforcement of the policy concerning the use and distribution of access codes. In addition, since the security modem simply verifies the location of a terminal by a callback over the PSTN, data

calls are still susceptible to illicit monitoring. Other limitations associated with security modems include the fixed telephone numbers stored in the modem's memory, the requirement and cost of a second call and the requirement of call originators to manually place some modems into an answer mode to receive the callback from the security modem.

Since a security modem requires the network administrator to associated fixed telephone numbers to access codes, this device is impractical for the traveling businessman to use. As an example of this, consider a salesperson who visits several customers during the day. Since a maximum of 20 access codes and associated telephone numbers can be entered into a security modem, even if the salesperson contacts the network administrator on a regularly scheduled basis it would be a demanding task to continuously adjust the security modem to correspond to one's sales calls.

The requirement of the security modem to initiate callbacks can considerably add to the cost of the organization's telephone bill. This is because the telephone companies in many metropolitan areas base the cost of a call upon message units. Since the initial call to the security modem results in a minimum of one message unit being used, even if the call only required 30 s for the user to enter his or her access code, message units can rapidly build up. For long-distance calls the cost associated with the requirement for an initial call and a callback may be more pronounced. This is due to the tariff structure of long-distance calls, where the cost of the first minute can be considerably higher than the cost per minute for succeeding minutes. As the initial call to the security modem is normally billed at the higher first-minute rate, this can considerably expand the communications cost of an organization over a period of time.

Another key limitation of security modems is the storage capacity of the device. If the modem, as an example, is limited to storing five access codes and associated telephone numbers a significant number of devices may be required to support a large number of users. As an example of this consider a geographical area that contains 50 terminals that will dial a multiplexer. If only six terminals are expected to be active at one time, six dial-in lines and conventional modems connected to the multiplexer would be sufficient. If security modems are required, 10 devices would be necessary, since each modem is assumed to store a maximum of five access codes and telephone numbers. This in turn would require 10 dial-in lines and 10 ports on the multiplexer, increasing the cost of communications facilities and equipment necessary when security modems are used. In addition, although a common rotary could be used with conventional modems, users dialing security modems would be restricted to accessing specific telephone numbers to which the modem with their access code is connected. This in turn further limits the flexibility associated with using security modems.

## 4.7 LINE DRIVERS

There are four basic methods used to connect a terminal device to a computer. These methods are listed in Table 4.30.

**Table 4.30** Terminal-to-computer circuit connections

Direct connection of terminals through the use of wire conductors
Connection of terminals through the utilization of line drivers
Connection of terminals through the utilization of limited distance modems
Connection of terminals through the utilization of modems or digital service units

It is interesting to observe that each of the methods listed in Table 4.30 provides for progressively greater distances of data transmission while incurring progressively greater costs to the users. In this section the limitations and cost advantages of the first two methods will be examined in detail.

## Direct connection

The first and most economical method of providing a data circuit is to connect a terminal device directly to a computer through the utilization of a wire conductor. Surprisingly, many installations limit such direct connections to 50 feet in accordance with terminal and computer manufacturers' specifications. These specifications are based upon EIA RS-232 and ITU V.24 standards. If the maximum 50-feet standard is exceeded, manufacturers may not support the interface, yet terminals have been operated in a reliable manner at distances in excess of 1000 feet from a computer over standard data cables. This contradiction between operational demonstrations and usage and standard limitations is easily explained.

If you examine the RS-232 and V.24 standards, such standards limit direct connections to 50 feet of cable for data rates up to 20 kbps. Since the data rate is inversely proportional to the width of the data pulses transmitted, taking capacitance and resistance into account, it stands to reason that slower terminals can be located further away than 50 feet from a computer without incurring any appreciable loss in signal quality. Simply stated, the longer the cable length the weaker the transmitted signal at its reception point and the slower the pulse rise time. As transmission speed is increased, the time beween pulses is shortened until the original pulse may no longer be recognized at its destination. This becomes more obvious when you consider that a set amount of distortion will affect a smaller (less wide) pulse than a wider pulse.

In Figure 4.66, the relationship between transmission speed and cable length is illustrated for distances up to approximately 3400 feet and speeds up to 40 000 bps. This figure portrays the theoretical limits of data transfer speeds over an unloaded length of 22 American wire gauge (AWG) cable. Many factors can have an effect on the relationship between transmission speed and distance including noise, distortion introduced owing to the routing of the cable, and the temperature of the surrounding area where the cable is installed. The ballast of a fluorescent fixture, for instance, can cause considerable distortion of a signal transmitted over a relatively short distance.

The diameter of the wire itself will affect total signal loss. If the cross-sectional area of a given length of wire is increased, the resistance of the wire to current
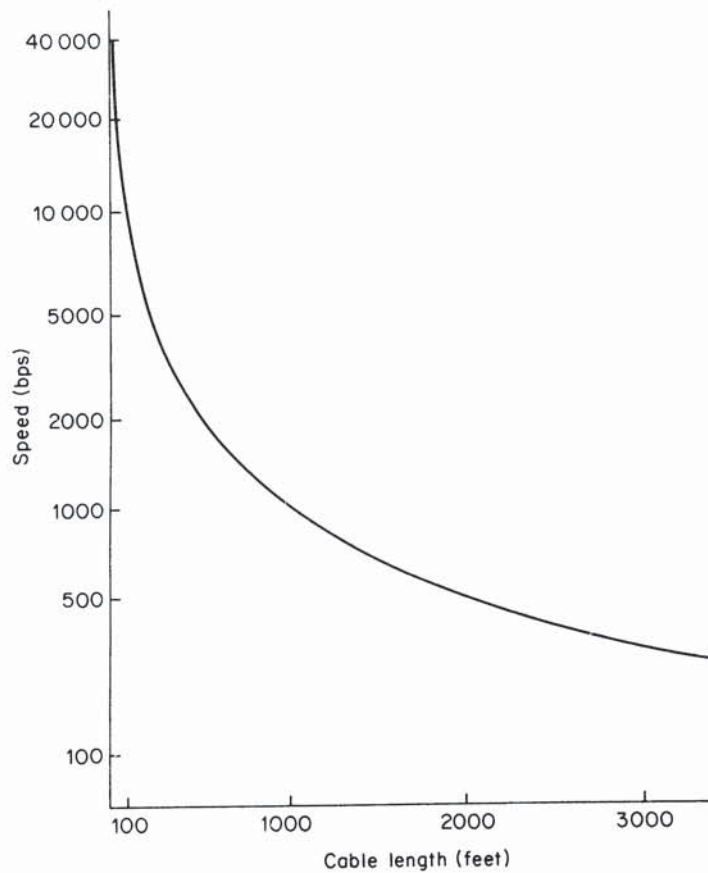
**Figure 4.66** Speed and cable length relationship

flow is reduced. Table 4.31 shows the relationship between the dimensions and resistances of several types of commercially available copper wire denoted by gauge numbers. By increasing the gauge from 22 to 19, the resistance of the wire is reduced by approximately one-half.

Another method which can be utilized to extend the length of a direct wire connection is limiting the number of signals transmitted over the data link. After the connect sequence or handshaking is accomplished, only two signal leads are required for data transfer: the transmitted data and receive data leads. With some minor engineering at both ends of the data link and an available dc voltage source, the remaining signals can be held continuously high, permitting the use of a simple paired cable to complete the data link.

Cable length can be further extended by the use of commercially available low-capacitance shielded cable. The shield consists of a thin wrapping of lead foil around the insulated wires and is quite effective in reducing the overall capacitance of the data cable with a very modest increase in price over standard unshielded cables. The use of low-capacitance shielded cables is strongly recommended when several cables must be routed through the same limited diameter conduit. Once the

**Table 4.31** Relationship between wire diameter and resistance

| Gauge number | Diameter (inch) | Ohms/1000 feet at 7°F |
|---|---|---|
| 10 | 0.102 | 1.02 |
| 11 | 0.091 | 1.29 |
| 12 | 0.081 | 1.62 |
| 13 | 0.072 | 2.04 |
| 14 | 0.064 | 2.57 |
| 15 | 0.057 | 3.24 |
| 16 | 0.051 | 4.10 |
| 17 | 0.045 | 5.15 |
| 18 | 0.040 | 6.51 |
| 19 | 0.036 | 8.21 |
| 20 | 0.032 | 10.30 |
| 21 | 0.028 | 13.00 |
| 22 | 0.025 | 16.50 |
| 23 | 0.024 | 20.70 |
| 24 | 0.020 | 26.20 |
| 25 | 0.018 | 33.00 |
| 26 | 0.016 | 41.80 |
| 27 | 0.014 | 52.40 |
| 28 | 0.013 | 66.60 |
| 29 | 0.011 | 82.80 |

practical limitation of cable length has been reached, signal attenuation and line distortion can become significant and either reduce the quality of data transmission or prevent its occurrence. One method of further extending the direct interface distance between terminals and a computer is by incorporating a line driver into the cable connection.

## Using line drivers

As the name implies, a line driver is a device which performs the function of extending the distance a signal can be transmitted down a line. A single line driver, depending upon manufacturer and transmission speed, can adequately drive signals over distances ranging from hundreds of feet up to a mile. One manufacturer introduced a line driver capable of transferring signals at a speed of 100 kbps at a distance of 5000 feet and a 1 Mbps signal over a distance of 500 feet using a typical multipair cable.

A multitude of names have been given to the various brands of line drivers to include local data distribution units and modem eliminator drivers. For the purpose of this discussion a line driver is a stand-alone device inserted into a digital transmission line in order to extend the signal distance.

In Figure 4.67 the distinction between single and multiple line drivers is illustrated and contrasted with limited distance modems which will be explored in Section 4.8. The primary distinction between line drivers and limited distance

**Single-line driver**

```
┌──────────┐      ┌────────┐                    ┌─────────┐
│          │      │ Line   │                    │         │
│ Computer │──────│ driver │────────────────────│ Terminal│
│          │      └────────┘                    └─────────┘
│          │
└──────────┘
```

Range in thousands of feet

**Multiple-line drivers**

```
┌──────────┐      ┌────────┐      ┌────────┐     ┌─────────┐
│          │      │ Line   │      │ Line   │     │         │
│ Computer │──────│ driver │──────│ driver │─────│ Terminal│
│          │      └────────┘      └────────┘     └─────────┘
│          │
└──────────┘
```

Range in thousands of feet

**Limited distance modems**

```
┌──────────┐    ┌──────────┐   ┌──────────┐    ┌─────────┐
│          │    │ Limited  │   │ Limited  │    │         │
│ Computer │────│ distance │───│ distance │────│ Terminal│
│          │    │ modem    │   │ modem    │    └─────────┘
│          │    └──────────┘   └──────────┘
└──────────┘
```

Range in miles
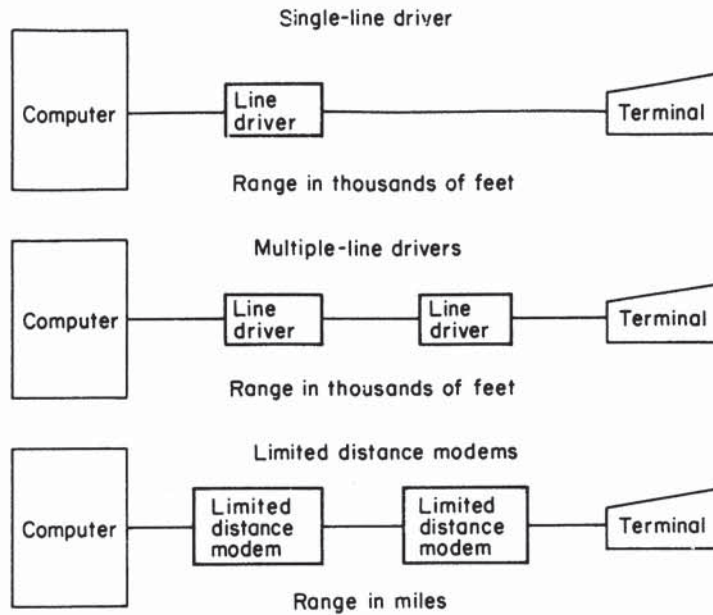
**Figure 4.67** Line drivers and limited distance modems. When line drivers are used, the signal remains in its digital form for the entire transmission. Distances can be extended by the addition of one or more line drivers which serve as digital repeaters. For longer distances, limited distance modems can be utilized where the transmitted data is converted into an analog signal and then reconverted back into its original signal by the modem

modems is that two identical units must be used as limited distance modems to pass data in analog form over a conductor, whereas a line driver serves as a repeater to amplify and reshape digital signals.

Although some models of line drivers can theoretically have an infinite number of repeaters installed along a digital path, the cost of the additional units as well as the extra cabling and power requirements must be considered. Generally, the use of more than two line drivers in a single digital circuit makes the use of limited distance modems a more attractive alternative.

## Applications

The characteristics of line drivers become important when one considers their incorporation into a data link. If the EIA RS-232 signals are accepted, amplified, regenerated, and passed over the same leads, they can be used as repeaters. If, however, the line driver also serves as a modem eliminator by providing a synchronous clock, inserting RTS/CTS delays and reversing transmit and receive signals, care should be taken when attempting to use them as repeaters. If this type of line driver is used and strapping options are provided for the RTS/CTS delay and such desirable features as internal/external clock, it is a simple matter to
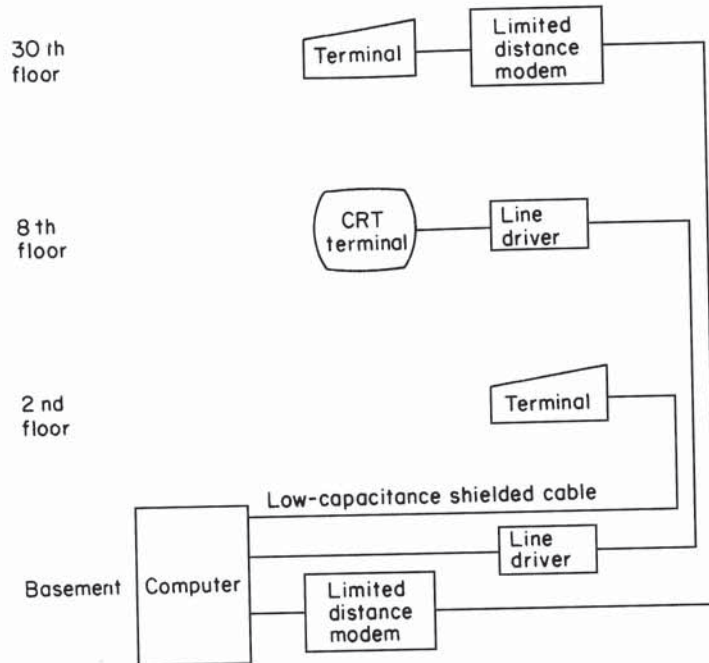
**Figure 4.68** A variety of methods can be used to connect a terminal to a computer located in the same building

convert it to a repeater by setting the delay to zero, setting the clock to external, and using a short pigtail cable to reverse the signals.

Figure 4.68 illustrates a typical application where line drivers would be installed. In this office building the computer system is located in the basement. The three terminals to be connected to the computer are located on different floors of the building. A remote terminal located on the second floor of the building is only 100 feet from the computer and is directly connected by the use of a low-capacitance shielded cable. The second terminal is located on the eighth floor, approximately 500 feet from the computer and is connected by the use of line drivers to extend the signal transmission range. A third terminal, located on the 30th floor, uses a pair of limited-distance modems for transmission since a large number of line drivers would be cost-prohibitive.

## 4.8 LIMITED-DISTANCE MODEMS

As the name implies, limited-distance modems are designed for data transmission over relatively short distances when compared to traditional modems. The utilization of such devices can result in dramatic savings in comparison to the cost of using conventional modems for the transmission of data over short distances. These devices have operating rates ranging from 110 bps to over 1 million bps for distances ranging from 1 to 20 miles or more. Names given to these

devices include not only limited-distance modems but such descriptive terms as modem bypass units, short-range data sets, short-haul modems, and wire line modems.

## Rationale and status

The rationale for the use of limited distance modems is one of economics. Since they are designed to transmit over relatively short distances, they do not require equalization circuitry and other features associated with more expensive analog leased line modems. Although the migration to the use of digital transmission facilities, coupled with a significant decrease in the cost of conventional modems, considerably reduced demand for limited distance modems, they still represent an economical method to interconnect devices within a building or campus area.

## Contrasting devices

In contrast to line drivers where one or more such devices are used to regenerate digital data and extend transmission ranges, limited distance modems require two matching components, one at each end of the circuit. In most cases, limited distance modems convert digital data into analog signals for transmission. Some devices on the market, however, convert a terminal's serial binary bits into bipolar return to zero signals to maintain transmission entirely in digital form. The utilization of this type of device can provide you with a direct-limited distance digital extension to a DATAPHONE® digital service/channel service unit (DSU/CSU).

## Transmission media

Some of the distances that can be achieved with limited-distance modems are illustrated in Figure 4.69 when transmission is over unloaded, twisted-pair metallic wire cables. The representative transmission distances illustrated represent a refined composite derived from manufacturers' data sheets and should be used as a guide to the variations in methods of signaling and sensitivity in receiving level between devices. As discussed in Section 4.7, the smaller the wire gauge the greater the diameter of the wire and the lower the resistance of the wire to passage of current. The four-wire gauges selected for illustration in Figure 4.69 are used because they correspond to the common wire sizes used in the US telephone systems, and many users prefer to utilize the existing common carrier cables in lieu of routing their own private cables. Of course, there is no gauge restriction in routing a private cable, and in light of the differences shown in Figure 4.69 between 19- and 26-gauge cable, many users will install a lower gauge cable even when transmitting at a normal rate to alleviate the necessity of recabling if the data transmission rate should increase at a later data.
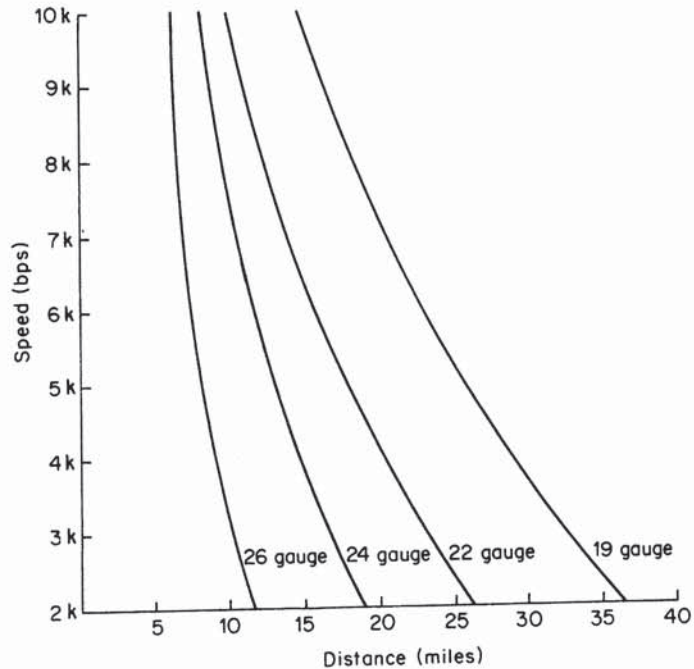
**Figure 4.69** Representative transmission distance—miles per twin pair wire gauges (unloaded)

**Table 4.32** Telephone wire sizes and resistances

| Circuit type | Gauge (AWG) | Diameter (mm) | Resistance 1000 ft | Resistance Loop mile |
|---|---|---|---|---|
| Station wire | 22 | 1.00 | 16 | |
| Station wire | 24 | 0.79 | 25 | |
| Station wire | 26 | 0.625 | 40 | |
| Toll wire | 19 | 1.42 | 8 | |
| Interoffice wire | 19 | 1.42 | 8 | |
| Open wire lines | | | | |
| Copper | 10 | 4.00 | | 6.7 |
| Copper | 12 | 3.00 | | 10.2 |
| Copper-clad steel | 12 | 3.00 | | 25 |
| Copper-clad steel | 14 | 2.00 | | 44 |

Common telephone wire sizes and associated measured resistances are listed in Table 4.32. In this table, a loop mile is a term which is used to describe two wires connecting two points which are physically located 1 mile apart, an important measurement when one considers using carrier facilities for supporting their limited-distance transmission requirements. If the use of existing carrier facilities is being contemplated, a close liaison with the local telephone company should be established. Common carriers and their operating companies in some areas may not

be completely familiar with this particular type of hardware or the tariff structure for the service to support limited-distance transmission. It is also important that the proposed limited-distance modem conforms to the specifications set forth in the Bell System publication 43401 entitled 'Transmission Specifications for Private Line Metallic Circuits'. This publication describes the signal level criteria objectives for private line metallic circuits (cable pairs without signal battery or amplification devices). In addition, the publication notes that the telephone companies have no obligation to provide private line channels on a metallic basis. Most manufacturers of limited-distance modems clearly specify that their equipment operate in accordance with the previously mentioned publication. If it is not explicitly stated, you may encounter delays and additional cost to ensure that the transmitter of the device is modified to comply with the specifications.

## Operational features

Most limited-distance modems utilize a differential diphase modulation scheme and permit internal transmit timing or external derived timing from the associated data terminal. These devices act similarly to a line driver, and most will accommodate four-wire half/full duplex and two-wire half-duplex/simplex data transmission. Data rate switches on some asynchronous units provide selectable data rates ranging from 110 to 28 800 bps, while the selectable rates of most synchronous units inlude various increments of 2400 bps. Users often select a transmission speed only to realize that by the time the equipment is installed changing requirements may indicate a different speed; therefore, the ease of adjustment of the unit should be investigated.

Some manufacturers state that their limited-distance modems can be inserted into a transmission line to serve as a repeater, as illustrated in Figure 4.70, to
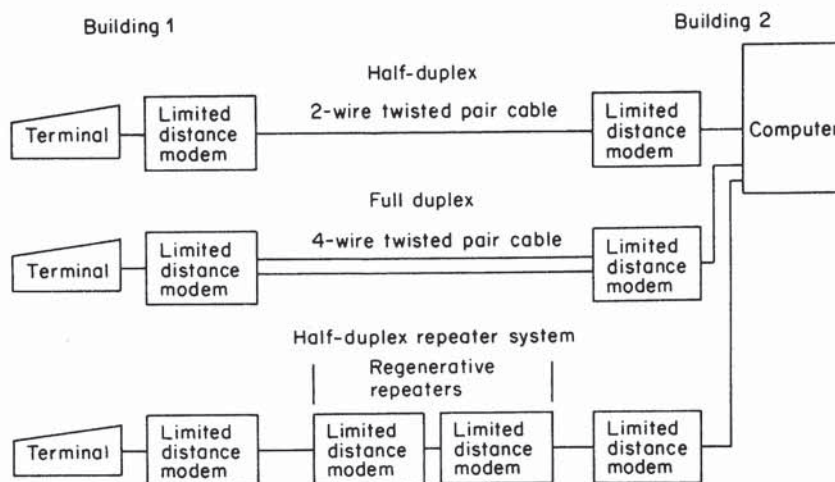


**Figure 4.70** Typical point-to-point applications. If a limited-distance modem is planned to be used as a repeater, care should be taken to ensure that shelter, a power source, and access for diagnostic testing and maintenance is available
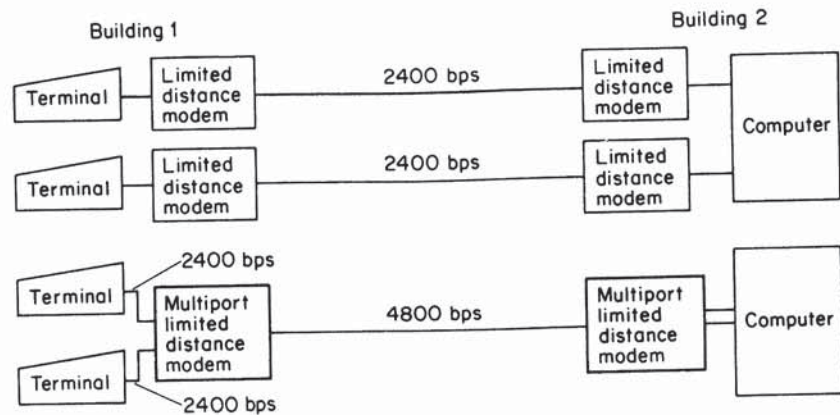
**Figure 4.71** Multiport operation reduces cabling requirements. Using a limited distance modem with a multiport feature, one cable may be utilized to provide access to a computer from several terminals

further extend the transmission distance. An obvious limitation to inserting such devices to serve as repeaters or data regenerators is the fact that they must be sheltered, have an available power source, and be readily accessible for diagnostics and maintenance. Since these combinations are difficult to achieve at locations between buildings, normal utilization of limited-distance modems has their locations fixed at each end of a transmission medium.

A very desirable feature that is offered on some units is a multiport or split stream feature. This feature permits several collocated terminals to utilize a single limited-distance transmission link at a considerable cost saving over the less expensive but more limited capability single-channel device. The key advantage to the employment of a multiport limited-distance modem is that only one cable instead of many cables can be used to service the transmission requirements of multiple terminals, as illustrated in Figure 4.71. For additional information on the advantages of multiport operations, the reader is referred to Section 4.4.

## Diagnostics

Diagnostic capabilities vary both by the model produced by a manufacturer and between manufacturers of these units. Some limited-distance modems have self-testing circuitry which permits the user to easily determine if the unit is operating correctly; or, if it is defective, it notifies the user of the operational status of the unit through the display of one or more lights which indicate equipment status and alarm conditions. Most self-testing features available with limited-distance modems involve some form of loopback testing. In such a test, the transmitter output of the modem is looped back or returned to the receiver of the same unit so that the transmitter signal can be checked for errors. Other tests which are available on some models include dc busback, in which the received data and clocking is transmitted back to the limited distance modem at the opposite end of the line to

provide an end-to-end test and a remote loopback test which can be used to trigger a dc busback at a remote location.

## 4.9 BROADBAND MODEMS

In this section we turn our attention to two types of modems that provide a high data transmission operating rate approximately 1000 times or more that provided by the fastest modem designed for use on the public switched telephone network. Earlier in this chapter I referred to such devices as cable and digital subscriber line (DSL) modems. That reference, which is still commonly used, denotes the primary application or intended application of each type of modem. That is, a cable modem is designed to provide data transmission capability on cable TV systems, while a so-called digital subscriber line modem is designed to support high-speed data transmission on telephone twisted-pair wiring into the home or office. Since the term 'broadband' is commonly used to represent a data transmission operating rate at or above 1.5 Mbps, which most cable and DSL modems support, I have taken author liberties and classified them collectively as *broadband modems*.

Both cable and DSL modems provide the potential to revolutionize communications technology as well as substantially to alter the manner by which we commonly perform daily communications-related activities at work and at home. Within a short period of time you may be able to surf the Internet's World Wide Web via the twisted-pair telephone line or a coaxial TV cable at operating rates 1000 or more times greater than those obtainable through the use of analog voice-grade modems, order video-on-demand movies via either type of media, and use either transmission system to communicate by voice, picture phone or perhaps even a technology requiring megabyte operating rates that will be developed tomorrow.

Currently, the dropping of regulatory barriers as well as the availability and affordability of megabyte transmission technology is fueling a race between cable TV operators and local telephone companies as they develop plans to enter each other's industry. Although a telecommunications reform bill was finally passed into law after years of debate, the actual implementation of various aspects of that law raises some questions concerning the degree of competition that will occur. However, it is with a degree of certainty that competition will occur through the use of cable and DSL modems that are the subject of this section. Thus, a good place to commence a discussion of broadband modems is by first reviewing the basic infrastructure of telephone and cable TV transmission, placing special emphasis on the methods used to connect subscribers to their backbone network via different local loop technologies.

## Telephone and cable TV infrastructure

The cabling infrastructure developed for telephone and cable TV (CATV) operations represents two diverse but to a degree potentially merging technologies. Telephone company plant is based upon the routing of twisted-pair wire from a

serving central office to individual subscribers to provide two-way communications. In comparison, cable TV was based upon the use of coaxial cable to provide a one-way transmission path for video distribution to subscribers.

### Telephone

Although the routing of wire-pairs directly to individual subscribers was long ago replaced in many areas by the multiplexing of several subscriber wire-pairs from a common location near groups of homes or offices to a telephone company central office as an economy measure, multiplexing maintains a direct path from the subscriber to the central office by either time or frequency, depending upon the technique employed. In addition, that path is bidirectional, enabling two-way voice and data to be transported from one subscriber to another via the telephone company wiring infrastructure.

Figure 4.72(a) provides a general schematic which illustrates the present telephone company wiring infrastructure. Although almost all long-distance transmission is carried via fiber optic cable, almost all local loop wiring is currently carried via metallic twisted-pair. Since the present twisted-pair local loop is limited by filters to a bandwidth under 3300 Hz, this also limits the maximum operating rate of voice-grade modems to approximately 33.6 kbps.

A second limitation involves the use of multiplexers that are referred to as a Digital Loop Carrier (DLC) system by a communications carrier. Although the use of a DLC system minimizes the length of local loops to subscribers, it places a limit on the actual bandwidth obtainable from a subscriber to a telephone company central office. This bandwidth limitation, as we will note later in this section, adversely affects the potential use of DSL modems and must be bypassed for the technology to operate.

Based upon several field trials conducted during the mid-1990s, telephone companies are beginning to expand the use of fiber by installing so-called 'fiber to the neighborhood'. Under this concept, fiber is routed from a central office to a common location in a housing subdivision, building complex, or similar site. Then, using electrical/optical converters the fiber trunk is connected to relatively short existing metallic twisted-pair wiring routed into homes and offices. Under this concept the high bandwidth of the fiber enables telephone companies to eliminate the filtering of twisted-pair local loops, and expand the bandwidth available per loop via multiplexing, which considerably expands the bandwidth available for use by modems. Thus, DSL modems can be used on the local loop to provide a much higher operating rate than that obtainable by the use of conventional twisted-pair wiring. Figure 4.72(b) illustrates the emerging telephone company wiring infrastructure.

Due to the considerable time and expense associated with recabling the existing telephone local loop infrastructure, several communications carriers are examining alternative methods to provide subscribers with high-speed transmission over existing local loop twisted-pair wire. Two promising techniques are known as High-Bit-Rate Digital Subscriber Line (HDSL) and Asymmetric Digital Subscriber Line (ADSL) which are described in detail later in this section. Although

(a) Current Infrastructure

```
        ┌──────────┐
        │ Central  │
        │ Office   │◄──
        │      ┌─┐ │
        │      │M│ │──
        └──────┴─┴─┘   \
             copper   ┌─┐
                  ────│M│────
                 /    └─┘    \
                ▼   ←copper→  ▼
              ┌─┐           ┌─┐
              │S│  • • •    │S│
              └─┘           └─┘
```
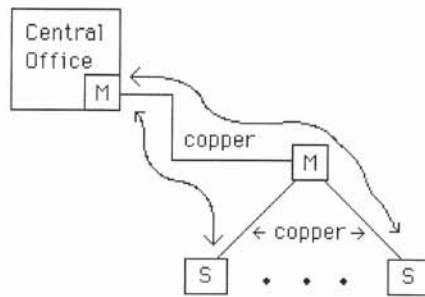
(b) Emerging Infrastructure

```
        ┌──────────┐
        │ Central  │
        │ Office   │◄──
        │    ┌───┐ │
        │    │E/O│ │──
        └────┴───┴─┘    \
            fiber    ┌───┐
                  ───│E/O│
                 /   └───┘
                │     ┌─┐
                │     │M│────
                │     └─┘    \
                ▼   ←copper→  ▼
              ┌─┐           ┌─┐
              │S│  • • •    │S│
              └─┘           └─┘
```
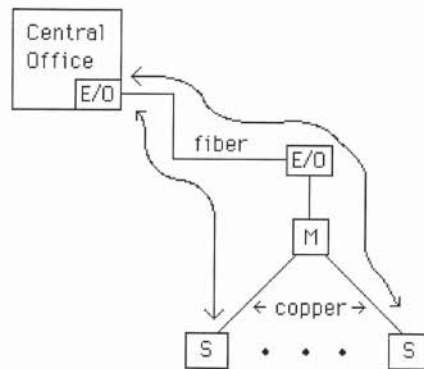
Legend:  | S | subscriber

| M | multiplexer

| E/O | electro-optic converter

**Figure 4.72** Current and emerging telephone company wiring infrastructure

HDSL and ADSL uses the existing twisted-pair local loop cable infrastructure, both require special equipment at the subscriber's premises and at the central office, the operation of which will be described later in this section.

## Cable TV

The cable television (CATV) infrastructure was originally developed to provide one-way video signals to subscribers. To do so, its wiring infrastructure was designed using a tree structure, with signals transmitted from a headend located at the beginning of the tree onto branches for distribution to subscribers. Signals on main branches were further split onto additional branches, with amplifiers installed

**(a)** Current Cable TV Wiring Infrastructure

**(b)** Emerging Hybrid Fiber/Coax

Legend:

FN   Fiber Node

S   Subscriber

A   Coax unidirectional amplifier

A   Coax bidirectional amplifier
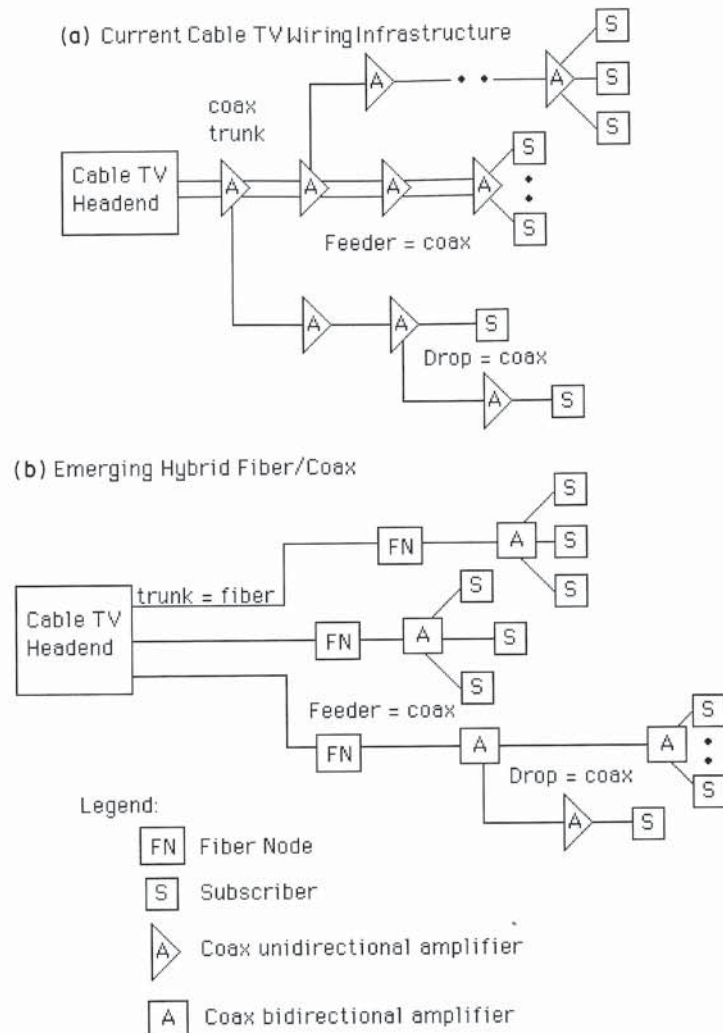
**Figure 4.73**   Current and emerging cable TV wiring infrastructure

to boost signal power since signal loss occurs due to cable distance as well as from the splitters encountered in the signal path.

The original bandwidth used on CATV systems ranged from 10 to 550 MHz, using 6 MHz per TV channel. Multiple signals are placed on a coaxial cable through the use of frequency division multiplexing equipment at the headend, with a tuner in subscriber set-top boxes used to extract the appropriate channel. This cabling and multiplexing scheme resulted in most CATV systems providing up to 83 channels to subscribers. The entire cable infrastructure of most pre-1995 CATV systems was based upon the use of coaxial cable as illustrated in Figure 4.73(a).

The CATV industry has had its share of trials to test different types of cabling and the support of two-way transmission by several competing techniques. One

technique involves the use of a second cable to provide a return path, while the second technique uses a different portion of the coaxial cable bandwidth to obtain a return path.

Both techniques involve a considerable change to the existing CATV cabling infrastructure. For example, using a different portion of the frequency spectrum for the return path requires the installation of bidirectional amplifiers, while the use of a second cable requires both a new set of amplifiers and cabling.

Recognizing that the use of the CATV cabling infrastructure could not accommodate the requirements of a large subscriber base for video-on-demand, high-speed data transmission, and even digitized telephone service, cable operators began to install hybrid fiber/coax (HFC) systems during the mid-1990s. Under HFC a star cabling infrastructure is used, with fiber cable routed from a cable TV switch to an optical distribution node commonly located in a subdivision or building complex. Then, coax is routed from the distribution node to individual subscribers. Under the HFC architecture downstream frequencies of 50 MHz to 750 MHz can be subdivided to provide a variety of analog and video channels to include data services and telephony. Upstream operations are limited to the 5 MHz to 40 MHz frequency band for telephony, data services, and control channels used by interactive video set-top boxes.

Figure 4.73(b) provides a general schematic of the evolving CATV HFC cabling infrastructure. Note that the amplifiers shown in Figure 4.73(b) must bidirectional to enable upstream transmission from subscribers to be supported. Table 4.33 compares the current and evolving operating characteristics of CATV and telephone compares transmission facilities. The evolving CATV transmission method in Table 4.33 requires a degree of elaboration. Currently one-way CATV systems broadcast frequency-multiplexed signals. While the evolving HFC infrastructure will still use a broadcast transmission method for delivery of basic cable video services, it will also employ switching technology to provide routing and delivery of voice, data, and video-on-demand in a manner similar to that by

**Table 4.33**  Cable TV (CATV) versus telephone operating characteristics

| Operating characteristic | CATV | Telephone |
|---|---|---|
| Bandwidth | 6 MHz/TV channel | 4 kHz/line |
| Use: | | |
|   Current | Video | Voice and data |
|   Evolving | Video, data, voice | Voice, data, Video |
| Directionality: | | |
|   Current | One-way | Two-way |
|   Evolving | Two-way | Two-way |
| Transmission method: | | |
|   Current | Broadcast | Switched |
|   volving | Broadcast/switched | Switched |
| Media: | | |
|   Current | Coaxial cable | Twisted-pair |
|   Evolving | Fiber + coaxial cable | Fiber + twisted-pair |

which the telephone company uses switching to establish a connection between a calling and a called party for a voice or data communications session. Now that we have an appreciation for the current and evolving cabling infrastructures used for CATV and telephone local loop operations, let's turn our attention to the broadband modems that will provide high-speed transmission on each cabling system.

## Cable modems

As an evolving technology there were several different engineering approaches being used to develop cable modems when this book was prepared. Although each approach results in an incompatibility between dissimilar vendor products, since it appears that each CATV operator will more than likely standardize their offering based upon a common vendor product, the inability of different cable systems to interoperate should not present a problem to subscribers. In fact, even if two cable systems use incompatible modems a subscriber on one system should be able to exchange electronic mail and operate other applications that require communications with subscribers on another system. This is true as long as the two systems are interconnected by a gateway and software on each system supports the application being used by each subscriber. This interoperability can be viewed as being similar to the Internet, where a person whose PC is attached to a LAN that in turn is connected to the Internet can communicate with a person who uses a modem to dial CompuServe, Prodigy, or America Online. In comparison, modem compatibility becomes an issue in the use of the switched telephone network where modem users are able to communicate with anyone connected to the PSTN as long as the called party has a compatible modem. Since CATV data operations can be expected to result in cable operators connecting the data portion of their network to commercial on-line services such as CompuServe and America Online as well as to the Internet, compatibility between cable modems manufactured by different vendors is not as important an issue as for modems developed for use on the PSTN. However, since standardization will make competition more meaningful for cable operators and vendors, a considerable amount of effort is presently being spent to standardize cable modem technology. One such effort is being made by the IEEE 802.14 Working Group under the title 'Standard Protocol for Cable-TV Based Broadband Communications Network'. Since the task of the IEEE 802.14 Working Group may not be completed for a considerable period of time, we will focus our attention on one announced product to obtain an appreciation for the operational capability of a cable modem.

### LANcity LCP

One of the first modem manufacturers to provide a city-wide data communications capability through the development of a series of products to include a cable modem was LANcity of Andover, Massachusetts, which was acquired by Bay Networks. Founded in 1990, by the summer of 1994 LANcity had introduced equipment to provide cable TV connectivity to the Internet. In April 1995,

LANcity introduced the world's fastest and least expensive personal cable TV modem, named LCP (LANcity Personal Cable TV Modem). This modem can be set to operate in any 6 MHz TV channel in the transmit range from 5 to 42 MHz and the receive frequency range of 54 to 750 MHz.

This modem uses Quadrature Phase Shift Keying (QPSK) modulation to obtain a symmetrical 10 Mbps operating rate, requiring approximately 5 MHz of RF bandwidth in a 6 MHz channel. Under QPSK the phase of the carrier signal is varied based upon the composition of the digital data to be transmitted. For example, a digital '1' could be transmitted by generating a 180° phase shift in the carrier, while a '0' might be represented by a 0° phase shift. The 'quadrature' aspect of the modulation scheme results in the carrier being capable of being shifted to one of four possible phases (0°, 90°, 180°, 270°) based upon the dibit value of the data to be transmitted. Figure 4.74 illustrates the four-point QPSK constellation pattern.
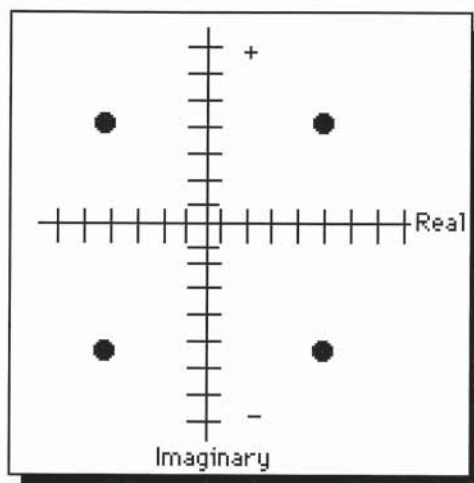


**Figure 4.74**  Four-point QPSK constellation

## Access protocol

Based on the method by which CATV channels are routed on the main trunk and feeder cables through drop cables to the subscriber, bandwidth is shared among subscribers. In comparison, a local loop twisted-pair connection from the telephone company central office to a subscriber represents dedicated bandwidth. This means that access to the 6 MHz channel the cable modem uses must be shared among many subscribers, requiring an access protocol to govern the orderly flow of data onto the cable. LANcity and most cable modem vendors support the Ethernet Carrier Sense Multiple Access/Collision Detection (CSMA/CD) protocol to provide a mechanism for cable subscribers to gain orderly access to the 6 MHz TV channel. To accomplish this the LANcity LCP cable modem is cabled to an Ethernet 10 Mbps adapter card installed in a personal computer. The PC uses software to establish a TCP/IP protocol stack which results in the computer

becoming in effect a workstation on a LAN that can represent up to 200 miles of CATV cable infrastructure.

In addition to requiring an Ethernet adapter card in each PC that will be connected to the cable modem, the upgrade of CATV plant to support data transmission via cable modems requires several additional components. These include a frequency converter, configuration server and router. Figure 4.75 illustrates their use on a CATV cabling infrastructure to provide subscribers with a bidirectional 10 Mbps operating capability using the LANcity LCP.
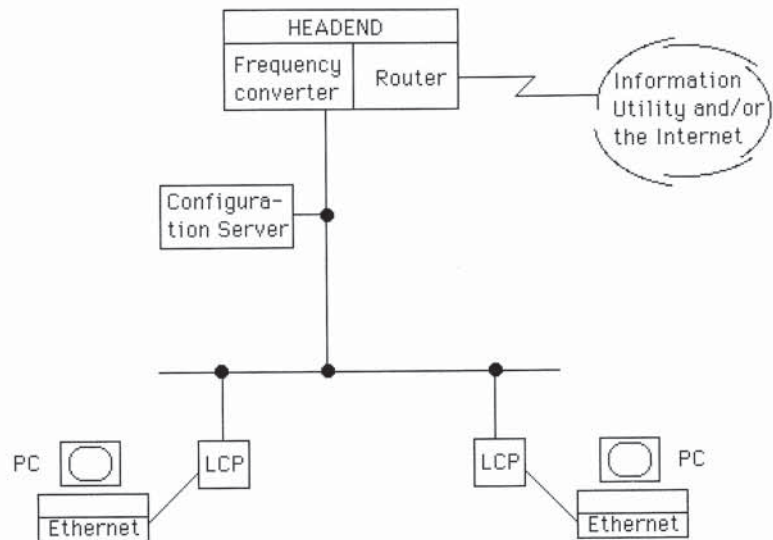


Legend:

PCP        LANcity Personal Cable TV Modem or similar product.

Ethernet  Ethernet adapter card installed in a personal computer.

**Figure 4.75**   CATV plant components supporting data transmission

## Frequency converter

The use of a frequency converter is required to change the reverse direction frequency to the forward direction frequency and vice versa. This conversion is required to enable one user to communicate with another as one channel provides a transmission path while a second channel is used for data reception. This conversion occurs at the headend and is limited to the 6 MHz channels used for data, ignoring other cable services. The LANcity product that performs frequency conversion is marketed as the LANcity TransMaster (LCT).

### Configuration server

The configuration server supports the TCP/IP network formed on the CATV cable infrastructure. In doing so the server performs address resolution services as well as managing IP addresses for the client computers using cable modems.

### Router

The function of the router is to encapsulate data into standardized frames for transport to external networks as well as to receive frames originating outside the CATV LAN.

### Operating rate comparison

To obtain an appreciation of the potential capability of a 10 Mbps modem, Table 4.34 provides a comparison of the file transfer time for a 10 Mbyte file at operating rates from 9600 bps (V.32 modem) to the LCP cable modem's 10 Mbps rate. Although the data in Table 4.34 indicates that the 10 Mbps operating rate of the LANcity LCP modem provides a significant reduction in file transfer time, it should be noted that the transfer time of 8 seconds is a mathematically derived time that will not be obtainable even under the best of circumstances. In actuality, a cable modem user will share the 10 Mbps transmission path to the headend with other users using the CSMA/CD access protocol. This means your actual throughput will be governed by the number of other subscribers transmitting data. In addition, the occurrence of Ethernet frame collisions results in each Ethernet adapter using a random exponential backoff algorithm to generate a waiting time prior to retransmitting a frame. Although this might increase file transfer time to 10 to 20 seconds, or even 30 or more on a heavily utilized LAN, it still represents a considerable improvement over the V.34 28 800 bps transfer time of 46 minutes.

**Table 4.34**  10 Mbyte file transfer time comparison

| Operating rate | Transfer time |
|---|---|
| 9600 bps (V.32 modem) | 138 minutes |
| 14 400 bps (V.32 bis modem) | 93 minutes |
| 28 800 bps (V.34 modem) | 46 minutes |
| 64 000 bps (ISDN B channel) | 20 minutes |
| 1544 000 bps (T1 DSU) | 52 seconds |
| 4000 000 bps (Token-Ring) | 20 seconds |
| 10 000 000 bps (Ethernet) | 8 seconds |

## IEEE 802.14 proposal

In a proposal to the IEEE 802.14 Working Group for the use of QAM for HFC downstream transmission, Scientific Atlanta noted that QAM is non-proprietary and was previously selected as the European Telecommunications Standard. In the

firm's proposal to the IEEE two levels of modulation based upon 64 QAM and 256 QAM were defined to permit implementation flexibility. Based upon the proposed modulation specifications, the standardization of QAM for downstream transmission would result in a signaling rate of 5 MHz using a carrier frequency between 151 MHz and 749 MHz spaced 6 MHz apart to correspond to TV channel assignments.

The use of a 5 MHz signaling rate and 64 QAM which enables six bits to be encoded in one signal change permits a transmission rate of 6 bits/symbol $\times$ 5 MHz, or 30 Mbps. In comparison, the use of 256 QAM results in the packing of eight bits per signal change, resulting in a transmission rate of 8 bits/signal change $\times$ 5 MHz, or 40 Mbps. Through the use of forward error coding, which is briefly described later in this section, the data rate throughput is slightly reduced from the modem's operating rate. This reduction results from extra parity bits becoming injected into the data stream to provide the forward error detection and correction capability.

## Constellation patterns

Figure 4.76 illustrates the signal constellation pattern for 64 QAM modulation while Figure 4.77 illustrates the pattern for one quadrant when 256 QAM modulation is used. Note that the signal element coding is differential quadrant coding using Gray coding within a quadrant. Under the Gray code the difference between two successive binary numbers is limited to one bit changing its state. Through the use of Gray code encoding the most likely error during demodulation in which an incorrect adjacent code is selected will result in a one-bit error when decoded at the receiver. Table 4.35 contains the binary and Gray code equivalent for 3-bit encoding.
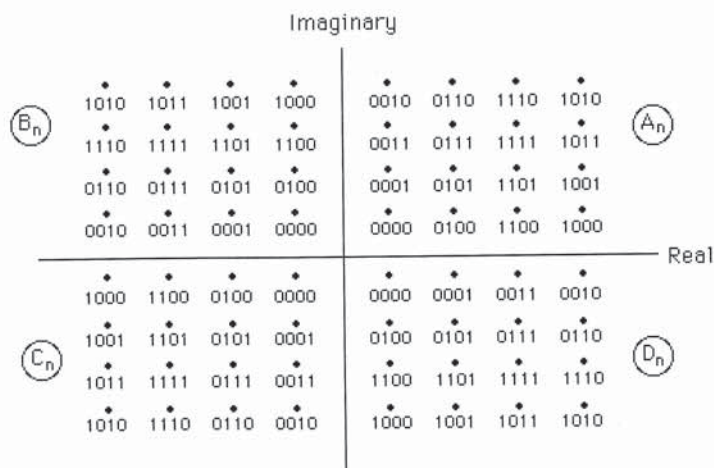


**Figure 4.76** Signal constellation pattern for 64 QAM modulation. The binary numbers denote b3$_n$, b2$_n$, b1$_n$, b0$_n$, and the letters A$_n$, B$_n$, C$_n$, and D$_n$ denote the four quadrants

Imaginary

```
000100 001100 011100 010100 110100 111100 101100 100100
  •      •      •      •      •      •      •      •
000101 001101 011101 010101 110101 111101 101101 100101
  •      •      •      •      •      •      •      •
000111 001111 011111 010111 110111 111111 101111 100111
  •      •      •      •      •      •      •      •
000110 001110 011110 010110 110110 111110 101110 100110
  •      •      •      •      •      •      •      •
000010 001010 011010 010010 110010 111010 101010 100010
  •      •      •      •      •      •      •      •
000011 001011 011011 010011 110011 111011 101001 100001
  •      •      •      •      •      •      •      •
000001 001001 011001 010001 110001 111001 101001 100001
  •      •      •      •      •      •      •      •
000000 001000 011000 010000 110000 111000 101000 100000
  •      •      •      •      •      •      •      •
```

$B_n$ ⊙                                                      ⊙ $A_n$

Real

$C_n$ ⊙   $D_n$ ⊙

**Figure 4.77**   Constellation pattern for 256 QAM modulation (one quadrant illustrated)

**Table 4.35**   Binary and Gray code equivalence

| Decimal | Binary | Gray Code |
|---------|--------|-----------|
| 0 | 000 | 000 |
| 1 | 001 | 001 |
| 2 | 010 | 011 |
| 3 | 011 | 010 |
| 4 | 100 | 110 |
| 5 | 101 | 111 |
| 6 | 110 | 101 |
| 7 | 111 | 100 |

For both 64 and 256 QAM element coding the outputs are encoded using a phase change based upon the inputs and previous inputs. Table 4.36 lists the proposed 64 and 256 QAM element coding.

In addition to proposing the use of 64 and 256 QAM for downstream operations, Scientific Atlanta recognized the need for error correction and detection due to the noise levels on broadband systems. In the Scientific Atlanta proposal a Reed Solomon Parity code was proposed using interleaving to provide protection against random as well as burst errors. Thus, the demodulated data rates of 30 Mbps for 64 QAM and 40 Mbps for 256 QAM would result in actual corrected data rates of 27.378 and 36.504 Mbps, respectively.

## DSL modems

As mentioned earlier in this section, a DSL modem derives its name from the fact that it was developed to support the transmission of data via copper twisted-pair wire on the line linking a subscriber to a central office. In actuality there is a family

**Table 4.36**  64 and 256 QAM element coding

| Inputs (b5$_n$b4$_n$) | Previous inputs | Phase change (degrees) | Outputs |
|---|---|---|---|
| A$_n$ | A$_{n-1}$ | 0 | 00 |
| A$_n$ | B$_{n-1}$ | 270 | 10 |
| A$_n$ | C$_{n-1}$ | 180 | 11 |
| A$_n$ | D$_{n-1}$ | 90 | 01 |
| B$_n$ | A$_{n-1}$ | 90 | 01 |
| B$_n$ | B$_{n-1}$ | 0 | 00 |
| B$_n$ | C$_{n-1}$ | 270 | 10 |
| B$_n$ | D$_{n-1}$ | 180 | 11 |
| C$_n$ | A$_{n-1}$ | 180 | 11 |
| C$_n$ | B$_{n-1}$ | 90 | 01 |
| C$_n$ | C$_{n-1}$ | 0 | 00 |
| C$_n$ | D$_{n-1}$ | 270 | 10 |
| D$_n$ | A$_{n-1}$ | 270 | 10 |
| D$_n$ | B$_{n-1}$ | 180 | 11 |
| D$_n$ | C$_{n-1}$ | 90 | 01 |
| D$_n$ | D$_{n-1}$ | 0 | 00 |

of DSL modems being developed, each having features and operating characteristics developed to support different transmission rates over copper twisted-pair wire for different transmission distances. Two of the more popular types of DSL modems are HDSL and ADSL, both of which we will examine in this section.

In our previous examination of modems we noted that their operating rate is limited to approximately 33.6 kbps due to the voice channel bandwidth of approximately 3300 Hz limiting modem transmission capacity. One method recognized many years ago to transmit at a higher operating rate was to remove loading coils on a subscriber line, in effect increasing the bandwidth on a twisted-pair wire to approximately 1 MHz. Doing so enables a modem that packs two bits per baud to obtain a data transmission rate of 2 Mbps, while one that packs four bits per baud theoretically has the potential to operate at 4 Mbps when a 1 MHz bandwidth is available for use. Unfortunately, we do not live in a perfect world and the use of Digital Loop Carrier (DLC) multiplexers, a bridge tap on a subscriber line as well as the attenuation of high frequencies served as constraints which limited development of DSL modems. Improvements in line coding techniques resulted in the development of High-bit-rate Digital Subscriber Line (HDSL) technology during the early 1990s as a substitute for T1 and E1 transmission facilities, while Asymmetrical Digital Subscriber Line (ADSL) technology was developed to provide a high-speed transmission capability into homes and offices.

## HDSL

HDSL represents a transmission technique which uses a dibit coding scheme referred to as 2 Binary, 1 Quaternary (2B1Q) that was originally developed for use with ISDN. Instead of transmitting on a single wire pair, HDSL uses two pairs

(four wires) to split the T1 transmission rate of 1.544 Mbps so that each wire pair operates at 784 kbps. Since transmission on each wire pair occurs at 784 kbps, using dibit coding results in the signaling (baud) rate being reduced to 392K, enabling transmission distances up to 12 000 feet to be obtained on 24 or 26 gauge wire without repeaters.

In the European area the first series of HDSL modems developed for E1 transmission at 2.048 Mbps required three wire pairs for a total of six wires to achieve a targeted subscriber line distance. More modern E1 HDSL implementations operate on two-pair copper wire, each providing an operating rate of 1.168 Mbps to achieve an aggregate transmission rate of 2.048 Mbps. Figure 4.78 illustrates the use of HDSL to provide an extended T1 or E1 transmission capacity. Note that the HTU-C is the mnemonic for an HDSL Transceiver Unit located in a carrier's central office while HTU-R represents the unit located at the subscriber's remote site. Both actually represent broadband transmission devices that operate over two-pair copper wire. Since conventional T1 and E1 transmission lines require the use of repeaters at approximately 6000 feet, the use of HDSL enables relatively long transmission distances to be obtained without having to construct shelter and provide power for repeaters on the loop.
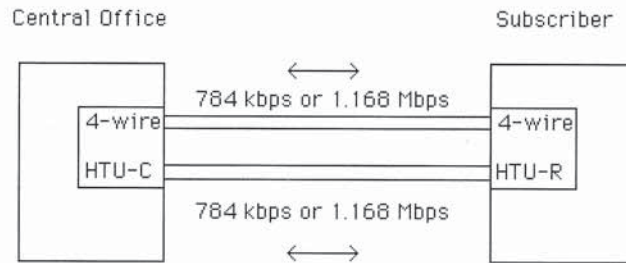


**Figure 4.78**   Using HDSL

## ADSL

Asymmetric Digital Subscriber Line (ADSL) represents a technology developed in recognition of the physical characteristics of frequency. That is, since high frequencies attenuate more rapidly than low frequencies, a larger band of frequencies is allocated to downstream transmission than to upstream transmission. While this reduces the data rate at which a subscriber can transmit towards a central office, it results in the received signal being relatively high in signal strength which reduces the effect of crosstalk at the central office. This in turn enables a greater subscriber line transmission distance to be achieved than if a symmetrical transmission method was employed.

A typical ADSL modem provides two 'data' channels, with the return channel having a maximum operating rate of 576 kbps, and results in the asymmetric operation of the modem. In actuality, an ADSL circuit with ADSL modems connected to each end of a twisted-pair line has three channels—a high-speed downstream channel, a medium-speed upstream channel, and a standard voice
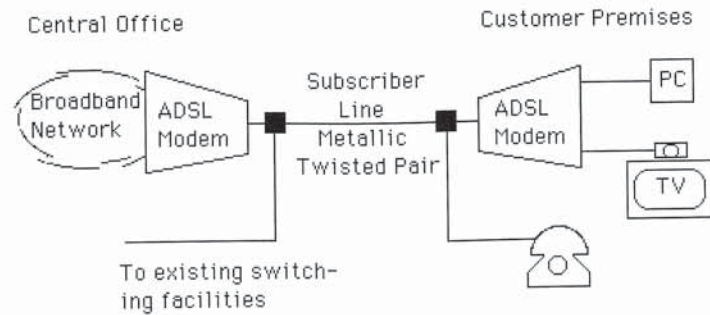
**Figure 4.79** Asymmetric Digital Subscriber Line. An ADSL supports three channels formed by frequency—a high-speed downstream channel, a medium-speed upstream channel, and a conventional telephone channel. The latter is formed through the use of filters

telephone channel. The latter is split off from the digital modems by filters, which ensures subscribers can continue to obtain the use of a voice telephone channel on the existing twisted-pair connection even if one or both ADSL modems fail.

Figure 4.79 illustrates the basic operation of an ADSL circuit. The downstream operating rate depends upon several factors, including the length of the subscriber line, its wire gauge, the presence or absence of bridged taps, and the level of interference on the line. Based upon the fact that line attenuation increases with line length and frequency, while it decreases as the wire diameter increases, we can note ADSL performance in terms of the wire gauge and subscriber line distance. Ignoring bridged taps which represent sections of unterminated twisted-pair cable connected in parallel across the cable under consideration, various tests of ADSL lines provided a general indication of their operating rate capability. That capability is summarized in Table 4.37.

**Table 4.37** ADSL performance

| Operating rate | Wire gauge | Subscriber line distance |
|---|---|---|
| 1.5/2.0 Mbps | 24 AWG | 18 000 feet |
| 1.5/2.0 Mbps | 26 AWG | 15 000 feet |
| 6.1 Mbps | 24 AWG | 12 000 feet |
| 6.1 Mbps | 26 AWG | 9 000 feet |

### Operation

ADSL operations are based upon advanced digital signal processing and the employment of specialized algorithms to obtain high data rates on twisted-pair telephone wire. Currently there are two competing technologies used to provide ADSL capabilities—Discrete Multitone (DMT) modulation and Carrierless Amplitude/Phase (CAP) modulation. The first technology, DMT, represents an American National Standards Institute (ANSI) standard. In comparison, CAP represents a proprietary technology developed by Paradyne Corporation, now an

independent company which was previously part of AT&T. At the time this book was prepared CAP had been licensed to a number of communications carriers throughout the world. Both DMT and CAP permit the transmission of high-speed data using Frequency Division Multiplexing (FDM) to create multiple channels on twisted-pair. Through the use of FDM the copper twisted-pair subscriber line is partitioned into three parts by frequency as illustrated in Figure 4.80. FDM assigns one channel for downstream data and a second channel for upstream data, while the third channel from 0 to 4 kHz is used for normal telephone operations. The downstream path can be subdivided through time division multiplexing to derive several high and low speed subchannels by time. In a similar manner the upstream channel can be subdivided.
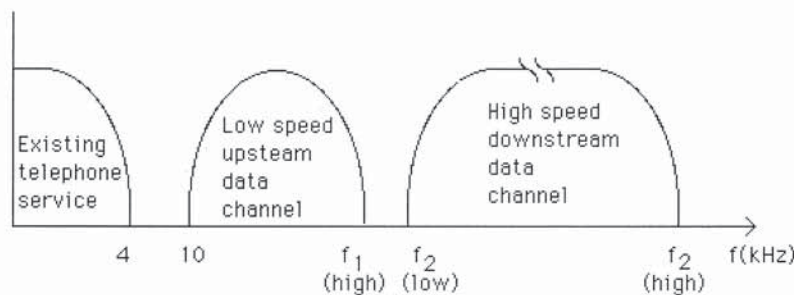
**Figure 4.80** ADSL frequency spectrum

## Discrete Multitone modulation

The concept behind Discrete Multitone (DMT) modulation is similar to that used in the Telebit Packetized Ensemble Protocol (PEP) modem described earlier in this chapter. That is, under DMT modulation available bandwidth is split or subdivided into a large number of independent subchannels. Since the amount of attenuation at high frequencies depends upon the length of the subscriber line and wire gauge, a DMT modem at the central office must determine which subchannels are usable. To do so that modem sends tones to the remote modem where they are analyzed. The remote modem responds to the central office modem subchannel scan at a relatively low speed which significantly reduces the possibility of the signal analysis performed by the remote modem being misinterpreted. Based upon the returned signal analysis, the central office modem will use up to 256 4-kHz wide subchannels for downstream transmission. Through a reverse measurement process the remote modem will use up to 32 4-kHz wide subchannels for upstream transmission.

One of the key advantages of DMT is its ability to take advantage of the characteristics of twisted-pair wire which can vary from one local loop to another. This makes DMT modulation well suited for obtaining a higher data throughput than is obtainable through the use of a single carrier transmission technique.

*Channel encoding*

Through the use of Quadrature Amplitude Modulation (QAM) up to 256 signal constellations are formed at 4 kHz intervals. Data to be transmitted is thus subdivided for encoding on separate channels and summed after demodulation at the receiver.

*Carrierless Amplitude/Phase (CAP) modulation*

Carrierless Amplitude/Phase (CAP) modulation is a derivative of QAM that was developed by Paradyne. Unlike DMT which subdivides the bandwidth of the wire into 4 kHz segments, CAP uses the entire bandwidth in the upstream and downstream channels. Under CAP serial data is encoded by mapping a group of bits into a signal constellation point using two-dimensional eight-state Trellis coding with Reed–Solomon forward error correction. The latter automatically protects transmitted data against impairments due to crosstalk, impulse noise, and background noise.

Figure 4.81 illustrates the CAP modulation process. Once a group of bits are mapped to a predefined point in the signal constellation, the in-phase and quadrature filters are used to implement the positioning in the signal constellation. Since this technique simply adjusts the amplitude and phase without requiring a constant carrier, the technique is referred to as 'carrierless'.
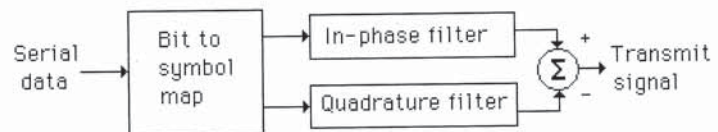


**Figure 4.81**   Carrierless Amplitude/Phase (CAP) modulation

The ADSL unit developed by Paradyne uses a CAP-256 line code (256-point signal constellation) for downstream operations, using bandwidth from 120 kHz to 1224 kHz. The composite signaling rate is 960 kbaud and seven bits are packed into each signal change to provide a downstream operating rate of 6.72 Mbps. However, the use of Reed–Solomon forward error correction reduces the actual payload to 6.312 Mbps plus a 64 kbps control channel. In the upstream direction the Paradyne device uses a CAP-16 line code in the 35 kHz to 72 kHz frequency band to obtain a composite signaling rate of 24 kbaud across 16 subchannels. Packing three bits per signal change, an upstream line rate of 72 kbps is obtained, of which 64 kbps is available for data.

Although cable and DSL modems are in their infancy when compared with more mature technologies, both types of devices hold the potential to revolutionize the manner by which we communicate. When we look to the past and think about how the video recorder and personal computer impacted our lives, we can also think about the future and plan for the coming revolution in communications that will occur through the use of broadband modems.

## 4.10 DIGITAL SERVICE UNITS

At the beginning of the 1970s, communications carriers began offering communication systems designed exclusively for the transmission of digital data. Specialized carriers, including the now-defunct DATRAN, performed a considerable service to the information-processing community through their pioneering efforts in developing digital networks. Without their advancements, major communications carriers may have delayed the introduction of an all-digital service.

In December 1974, the FCC approved the Bell System's DATAPHONE® digital service (DDS), which was shortly thereafter established between five major cities. Since then the service has been rapidly expanded to the point where, by 1998, more than 100 cities had been added to the DDS network.

Since the initial offering of DDS by AT&T, most communications carriers in the US and Europe have tariffed equivalent facilities. Today you can use digital transmission facilities from MCI, Sprint, and British Telecom, as well as from many other carriers.

### Comparison of facilities

With the voice-grade type of analog transmission, the data travels in a continuous manner; although it is easily amplified, any noise or distortion along the link is also amplified. In addition, the data signals become highly attenuated or weakened by the telephone characteristics originally geared to voice transmission. For the analog transmission of data, expensive and complex modems must be employed at both ends of the link to shape (modulate) and reconstruct (demodulate) the digital signals.

When digital transmission facilities are used, the data travels from end to end in digital form with the digital pulses regenerated at regular intervals as simple values of one and zero. Inexpensive digital service units are employed at both ends of the link to condition the digital signals for digital transmission.

AT&T's DDS is strictly a synchronous facility providing full-duplex, point-to-point, and multipoint service limited to speeds of 2.4, 4.8, 9.6, 19.2 and 56 kbps. In addition to these leased-line services, AT&T introduced a switched 56 kbps digital service in 1985 as well as switched 384 kbps and 1.544 Mbps services that were in limited use when this book was prepared. Access to AT&T's Switched 56 service is obtained by dialing a '700' number, which was available in over 100 US cities in early 1998.

Rates for leased-line digital services are based primarily upon distance and transmission speeds. This type of digital service is normally cost effective for high-volume users that could justify the expense associated with a dedicated communications facility. In comparison, pricing for switched 56 kbps service is usage sensitive, based on both connection duration and distance between calling and called parties. Due to the cost of this service essentially corresponding to usage, it is attractive for such applications as the backup of critical DDS and T1 lines, peak-time overload usage to eliminate the necessity of installing additional leased

digital circuits, and infrequent activities that may require a high data rate, such as still-frame and full-motion videoconferencing and facsimile transmission.

Terminal access to the DDS network is accomplished by means of a digital service unit which alters serial unipolar signals into a form of modified bipolar signals for transmission and returns them to serial unipolar signals at the receiving end. The various types of service units will be discussed in detail later in this section.

## Digital signaling

One of the most critical issues to be addressed in the design of digital transmission facilities is the method by which binary data will be encoded as signal elements for transmission. The selection of one encoding method over another affects both the cost of constructing transmission facilities and the resulting quality of transmission obtained from the use of such facilities.

As previously indicated in Section 1.3, there are a number of digital encoding techniques communications carriers considered prior to selecting Alternate Mark Inversion (AMI) with a 50% duty cycle. Its selection eliminated high frequency components of a signal that could interfere with other transmissions, and concentrated power in the middle of the transmission bandwidth which minimized distortion. In addition, as a bipolar signal it prevents dc voltage buildup, enabling transformer coupling which in turn permits both power and signal to be carried over a common cable, and separated to power repeaters on a span line. Although AMI was well suited for its intended purpose, it is a very simple modulation technique which uses unibit encoding, where one bit is packed into each signal change. This means that a T1 circuit operating at 1.544 Mbps requires the use of the frequency spectrum from 0 to 1.544 MHz, resulting in a maximum transmission distance of 6000 feet prior to requiring a repeater to rebuild the signal. Although AMI is still widely deployed, its transmission limitation resulted in the development of HDSL previously described in Section 4.9.

When using a bipolar transmission method such as AMI, a bit error occurs in the form of a bipolar violation. Thus, let's turn our attention to this topic.

### Bipolar violations

Bipolar transmission requires that each data pulse representing a logical one is transmitted with alternating polarity. A violation of this rule is defined as two successive pulses that have the same polarity and are separated by a zero level.

A bipolar violation indicates that a bit is missing or miscoded. Some bipolar violations are intentional and are included to replace a long string of zeros that could cause a loss of timing and receiver synchronization or to transmit control information. Figure 4.82 illustrates one example of a bipolar violation. Figure 4.82(a) shows the correct encoding of the bit sequence 010101010 using bipolar return to zero signaling. In Figure 4.82(b), the third 1 bit is encoded as a negative pulse and represents a violation of the bipolar return to zero signaling technique

(a)
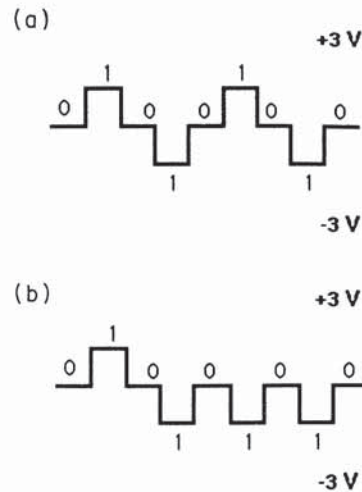
Figure 4.82 Bipolar violations. Two successive negative or positive pulses represents a bipolar violation of a bipolar return-to-zero signaling technique: (a) bipolar coding of data; (b) bipolar violation

where 1s are alternately encoded as positive and negative voltages for defined periods. In Section 4.11, we will examine several methods used to develop bipolar violations that are used to maintain synchronization when a string of consecutive zeros is encountered. These methods are commonly referred to as zero suppression codes.

## DDS structure

DDS facilities are routed from a subscriber's location to an Office Channel Unit (OCU) located in the carrier's serving central office. Since there is a variety of multiplexing methods that can be employed by a communications carrier to combine DDS facilities onto a T-carrier, let us focus our attention upon two methods that will illustrate the relationship between DDS and a T1 circuit. Figure 4.83 illustrates the multiplexing arrangement within an AT&T serving central office that supports DDS transmission at 9.6 kbps and 56 kbps.

The data service units (DSUs) at the subscriber's location can be viewed as 'digital modems' since they modulate the unipolar signal received from data terminal equipment, including computer ports, multiplexer ports and terminal ports, into a modified bipolar signal suitable for transmission on the DDS network. Originally, separate channel service units (CSUs) and data service units were required to interface equipment to the DDS network. Today, most vendors manufacture DSUs that, in effect, combine the functions performed by separate DSUs and CSUs.

The user TDMs shown in Figure 4.83 illustrate two methods by which end-users can transmit data to maximize the data handling capacity of different DDS facilities. The user TDM shown in the upper left corner of Figure 4.83 illustrates
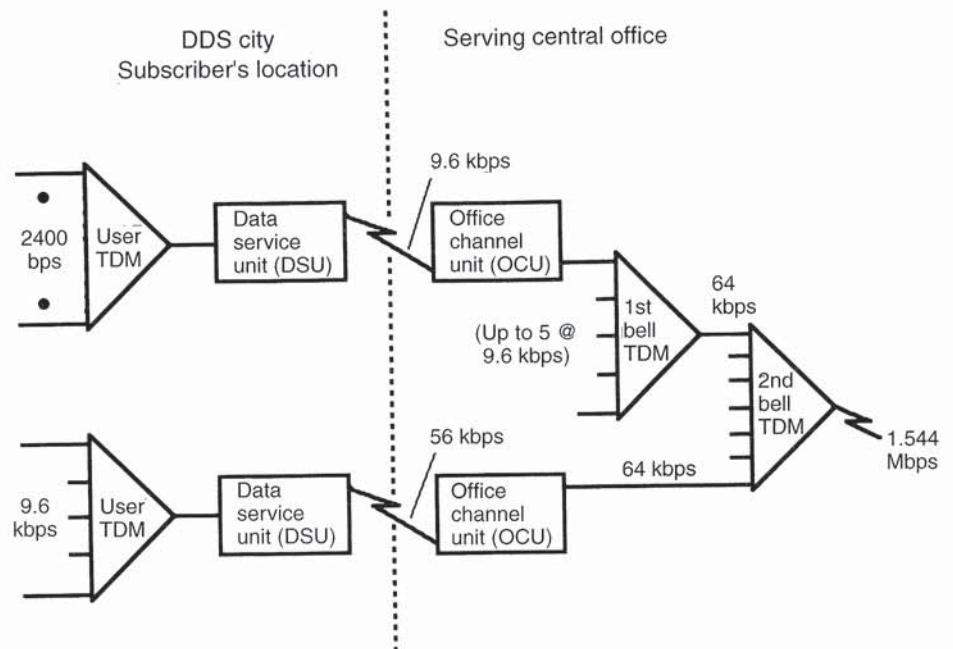
**Figure 4.83**  DDS multiplexing arrangement

how asynchronous transmission can be supported on DDS which is an all-synchronous transmission facility. In this example, 2400 bps asynchronous data sources are multiplexed into a 9.6 kbps synchronous data source for transmission onto DDS via the use of a DSU operating at 9.6 kbps. In the lower left portion of Figure 4.83, five 9.6 kbps asynchronous or synchronous data sources are multiplexed to obtain a 56 kbps synchronous data rate suitable for transmission on DDS. In both examples, one physical DDS circuit is used to transmit multiple logical channels of data.

The signals from the DSUs are terminated into a complementary office channel unit in the serving central office. From there, they enter into a multiplexing hierarchy which may carry voice as well as data.

## Framing formats

One of the more interesting aspects of DDS is the constraints upon its transmission rate resulting from the formats used to encode user data. User data transmitted at 56 kbps is increased to a DS0 64 kbps data rate at the OCU, and that device inserts groups consisting of seven bits of customer data into an 8-bit byte as illustrated in Figure 4.84(a). In this encoding format the control bit (C) added to every seven bits of customer data is set to a '1' if the byte contains customer data, while a value of '0' indicates that the byte contains network control data, such as idle or maintenance codes or control information. Since a DS0 signal results in the transmission of an 8-

(a) | $D_1$ | $D_2$ | $D_3$ | $D_4$ | $D_5$ | $D_6$ | C |

(b) | F | $D_1$ | $D_2$ | $D_3$ | $D_4$ | $D_5$ | $D_6$ | C |

(c) **Byte 1** | 0 | P & EC | C |

**Byte 2** | 1 | $D_1$ | $D_2$ | $D_3$ | $D_4$ | $D_5$ | $D_6$ | C |

**Byte 3** | 1 | $D_7$ | $D_8$ | $D_9$ | $D_{10}$ | $D_{11}$ | $D_{12}$ | C |

**Byte 4** | 0 | P & EC | C |

**Byte 5** | 0 | P & EC | C |
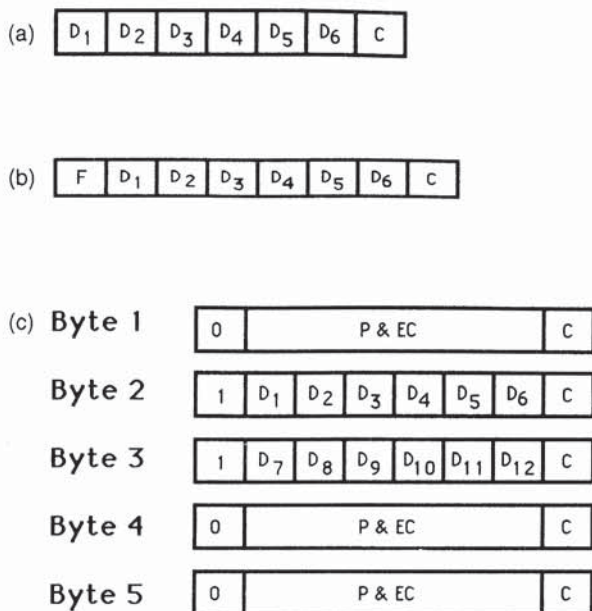
**Figure 4.84** DDS framing formats: (a) 56 kbps; (b) 2.4, 4.8, 9.6 kbps; (c) 19.2 kbps (F frame bit, D data bits, C control bit, P&EC parity and error correction bits)

bit byte 8000 times per second, this framing format results in 8 kbps of control bits being added to the 56 kbps customer data rate.

The construction of DS0 signals from the 2.4, 4.8 and 9.6 kbps DDS subrates is illustrated in Figure 4.84(b). As indicated, customer data is inserted into 8-bit bytes with six bits of user data framed by a frame bit (F) and a control bit (C). Once 2.4, 4.8 or 9.6 kbps DDS data streams are framed, one of two methods is used to place the framed data onto a DS0 channel. When 'byte stuffing' is used, the frame bit is set to '1' and the customer data are repeated the required number of times to create a 64 kbps DS0 signal. Thus, the 8-bit byte containing six bits of user data is repeated at 5, 10, and 20 times to enable 9.6, 4.8 and 2.4 kbps DDS data to be placed on a 64 kbps channel. When the F bit is set to '1' the frame format illustrated in Figure 4.84(b), is referred to as a DS0-A format. Thus, DS0-A data can bypass the first level TDM illustrated in Figure 4.83 and be fed directly into the second level TDM.

The second method of placing 2.4, 4.8 or 9.6 kbps DDS data onto a DS0 channel involves the use of the first level TDM illustrated in Figure 4.83. When this occurs, five 9.6, ten 4.8, or twenty 2.4 kbps formatted signals are multiplexed onto a single DS0 channel. To distinguish between the repeating of the same data resulting from byte stuffing and the multiplexing of different DDS signals, the framing bit is altered from all ones in byte stuffing to a subrate framing pattern to indicate multiplexing of different DDS data sources. When this framing pattern occurs, the resulting framing format is referred to as a DS0-B format. Obviously, DS0-B

formatting is more efficient than DS0-A formatting as the latter would require 20 DS0 channels to transmit twenty 2.4 kbps signals, while the former would require only one DS0 channel.

The introduction of 19.2 kbps DDS service required a substantial framing format change to accommodate this data rate. As illustrated in Figure 4.84(c), five bytes are required to carry 19.2 kbps customer data since three bytes are used for parity and error correction functions. In this framing format the frame bit in each byte (bit 1) results in a '01100' repeating pattern. Twelve bits of customer data are placed into two six-bit groups contained in bytes 2 and 3, resulting in 12 data bits being carried in every five-byte group of 40 bits. Thus, the use of the 64 kbps DS0 channel produces an effective data rate of $64 \times 12/40$, or 19.2 kbps.

### Signaling structure

A modified bipolar signaling structure is used on DDS facilities. The modification to bipolar return to zero signaling results in the insertion of zero suppression codes to maintain synchronization whenever a string of six or more zeros is encountered. Otherwise, repeaters on the span line routed between the carrier office and the customer may not be able to obtain clocking from the signal and could then lose synchronization with the signal.

To ensure a minimum ones density, at 2.4, 4.8, 9.6, and 19.2 kbps any sequence of six consecutive zeros is encoded as $000X0V$, where

0 denotes 0 V transmitted (binary 0)
$X$ denotes a zero or $+$ or $-A$ V, with the polarity determined by conventional bipolar coding
$V$ denotes $+$ or $-A$ V, with the polarity in violation of the bipolar rule.

Figure 4.85 illustrates the zero suppression sequence used to suppress a string of six consecutive zeros. For transmission at 56 kbps, any sequence of seven consecutive zeros is encoded as $0000X0V$.
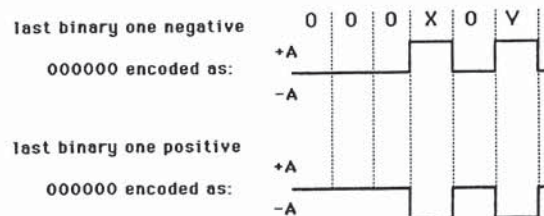


**Figure 4.85** DDS zero suppression sequence

*Timing*

Precise synchronization is the key to the success of an all-digital network. Timing ensures that data bits are generated at precise intervals, interleaved in time and read out at the receiving end at the same interval to prevent the loss or garbling of data.

To accomplish the necessary clock synchronization on the AT&T digital network, a master clock is used to supply a hierarchy of timing in the network. Should a link to the master clock fail, the nodal timing supplies can operate independently for up to two weeks without excessive slippage during outages. In Figure 4.86, the hierarchy of timing supplies as linked to AT&T's master reference clock is illustrated. As shown, the subsystem is a treelike network containing no closed loops.
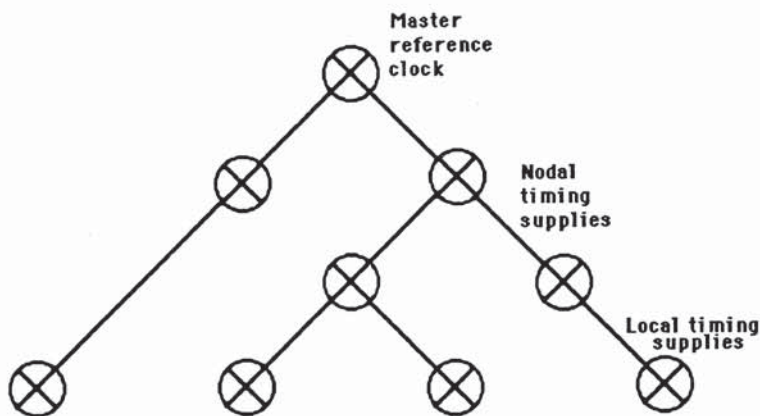


**Figure 4.86**   DDS timing subsystem

## Service units

When DDS was introduced, both a channel service unit (CSU) and a data service unit (DSU) were required to terminate a DDS line.

The DSU converts the signal from data terminal equipment into the bipolar format used with DDS and T1 facilities.

The DDS master reference clock is an atomic clock that is accurate to 0.01 part per million (PPM). This clock was installed by AT&T at Hillsboro, Missouri, which is the geographic center of the United States and whose location ensures a minimal variance in propagation delay time between DDS nodes connected to the master reference clock. The master reference clock oscillates at a rate known as the basic system reference frequency and is the most accurate of three timing sources used by digital facilities. The other two sources of timing include channel banks and loop timing where clocking is obtained from a high speed circuit.

The DSU interface to the DTE is accomplished by the use of a standard 25 pin EIA RS-232/V.24 female connector on the 2.4 kbps to 19.2 kbps units. The

**Table 4.38**  DSU interchange circuits

| RS-232 interface (DCE) | | V.35 interface (DCE) | |
|---|---|---|---|
| Pin | Signal | Pin | Signal |
| 1 | Chassis ground | P | Transmit data (A) |
| 2 | Transmit data | S | Transmit data (B) |
| 3 | Receive data | R | Receive data (A) |
| 4 | Request-to-send | T | Receive data (B) |
| 5 | Clear-to-send | C | Request-to-send |
| 6 | Data set ready | D | Clear-to-send |
| 7 | Signal ground | H | Data terminal ready |
| 8 | Carrier detect | E | Data set ready |
| 9 | Positive voltage | B | Signal ground |
| 10 | Negative voltage | F | Receive line signal detect |
| 15 | Transmit clock | Y | Transmit timing (A) |
| 17 | Receive clock | AA | Transmit timing (B) |
| 20 | Data terminal ready | V | Receive timing (A) |
| 24 | External transmit clock | X | Receive timing (B) |
| | | U | External transmit timing |
| | | W | External transmit timing |
| | | L | Local loopback |

wideband, 56 kbps device utilizes a 34 pin ITU, V.35 (Winchester) female type connector. Table 4.38 lists the RS-232 and V.35 interchange circuits commonly used by most DSUs. Since DTEs are normally attached to the DSU, the latter's interface is normally configured as data communications equipment (DCE) by the manufacturer.

Prior to deregulation, the CSU was provided by the communications carrier, while the DSU could be obtained from the carrier or from third-party sources. This resulted in an end-user connection to the DDS network similar to that illustrated in the top portion of Figure 4.87 where the CSU terminated the carrier's four-wire loop and the DSU was cabled to the CSU. In this configuration the CSU terminates the carrier's circuit. In addition, a separate CSU was designed to perform signal regeneration, monitor incoming signals to detect bipolar violations and perform remote loopback testing. Interfacing between the DSU and CSU is accomplished by the use of a 15-pin female D-type connector which utilizes the
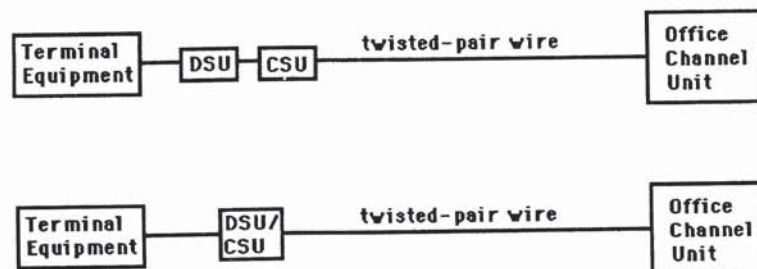


**Figure 4.87**  DSU/CSU connection

first six pins, where pin 1 is signal ground, pin 2 is status indicator, pins 3 and 4 are the receive signal pair, while pins 5 and 6 are the transmit signal pair. Today most communications carriers and third-party vendors manufacture combined DSU/ CSU devices, integrating the functions of both devices into a common housing which is powered by a common power supply. The lower portion of Figure 4.87 illustrates the connection of end-user terminal equipment to DDS using a combined DSU/CSU unit.

## DSU/CSU tests and indicators

Through the use of intentional bipolar violations, the DSU/CSU can generate a request to the OCU for the loop-back of the received signal onto the transmit circuit or it can interpret DDS network codes and illuminate relevant indicators on the device. When the loop-back button on the DSU/CSU is pressed, the device will generate four successive repetitions of the sequence $0B0X0V$ when operating at data rates up to 19.2 kbps or $N0B0X0V$ at 56 kbps, where

$B$ denotes $+$ or $-A$ V, with the polarity determined by bipolar coding for a binary 1
$X$ denotes a 0 V for coding of binary 0 or $B$, depending upon the required polarity of a bipolar violation
$V$ denotes $+$ or $-A$ V, with the polarity determined by the coding of a bipolar violation
$N$ denotes a do not care condition where the coding for a binary 0 or binary 1 is acceptable

In Figure 4.88 the DDS loop-back 6-bit sequence is illustrated for data rates at or under 19.2 kbps. Note that the sequence transmitted is dependent upon whether the previous binary 1 was transmitted as a positive or negative voltage.

Other bipolar violation sequences used by DDS include an idle sequence which indicates that a DTE does not have data to transmit, an out-of-service sequence and an out-of-frame sequence. The idle sequence is generated by the DSU while the out-of-service and out-of-frame sequences indicate a problem in the DDS network
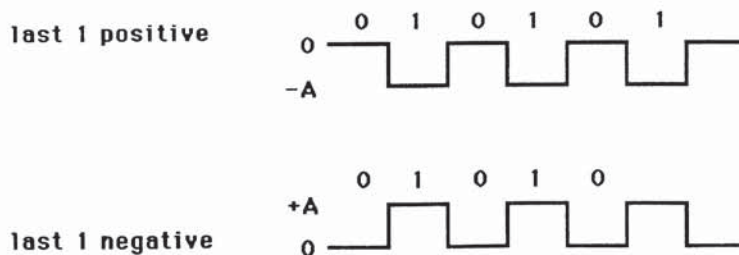


**Figure 4.88** DDS loop-back sequence (data rates up to 19.2 kbps). DDS loop-back codes are intentional bipolar violations

and are generated by the network and used by the DSU to illuminate an appropriate indicator on the device.

## DDS II

During 1988, AT&T introduced a new version of its DDS facility commonly referred to as DDS II. One of the key advantages of DDS II is its capability to provide a diagnostic channel along with the primary subrate channel. This diagnostic channel is obtained through a modification to the framing used by DDS and requires the use of special DSU/CSU that support the new channel.

Through the use of DDS II, end-users can perform non-disruptive testing or use the channel for network management purposes. The key to obtaining the ability to derive a secondary channel on DDS is the use of the network control or C bit.

In conventional DDS the C bit, which is bit eight in each DDS 8-bit byte, is transmitted as a binary one whenever a DTE requests access to a channel by turning its request to send (RTS) signal on. With the C bit continuously set to a one the DTE can transmit an unrestricted stream of data to include continuous zeros since every eighth bit will be automatically set to a one. By robbing this bit once every third byte, AT&T established a virtual path for diagnostic use.

The diagnostic channel data rate for 56 kbps DDS II is obtained by multiplying the full DS0 rate of 64 kbps by 1/8 which represents the C bit's portion of the DS0 rate to obtain 8 kbps. Next, since the bit robbing occurs every third byte, the resulting data rate becomes $8000 \times \frac{2}{3}$, or $2666 \frac{2}{3}$ bps. Similarly, dividing 2666 2/3 bps by the number of 19.2, 9.6, 4.8 or 2.4 kbps channels multiplexed onto a DS0 channel results in the diagnostic data rate for DDS II at those data rates.

AT&T added diagnostic channel capability throughout the DDS network. Although older DSU/CSU devices can support transmission on DDS II, those devices cannot support the use of the secondary channel capability provided by this modification to DDS. To do so requires the use of newer DSU/CSU devices that support the multiplexing of diagnostic data onto every third C bit position.

## Analog extensions to DDS

AT&T provides an 831A data auxiliary set which allows analog access to DDS for customers located outside the DDS servicing areas. The 831A connects the EIA RS-232 interfaces between a data service unit (500A-type) and a voice-band data set or modem. The 831A contains an 8-bit elastic store, control, timing, and test circuits which allow loop-back tests toward the digital network. The elastic store is a data buffer that is required by the DSU to receive data from the modem in time with the modem's receive clock. The data is then held in the elastic store until the DSU's transmit clock requests it. Thus, the buffer serves as a mechanism to overcome the timing differences between the clocks of the two devices. In the reverse direction, no buffer is required when the DSU's receive clock is used as the modem's external transmit clock. When the modem cannot be externally clocked or when one DSU is connected to a second DSU or a DTE that cannot accept an
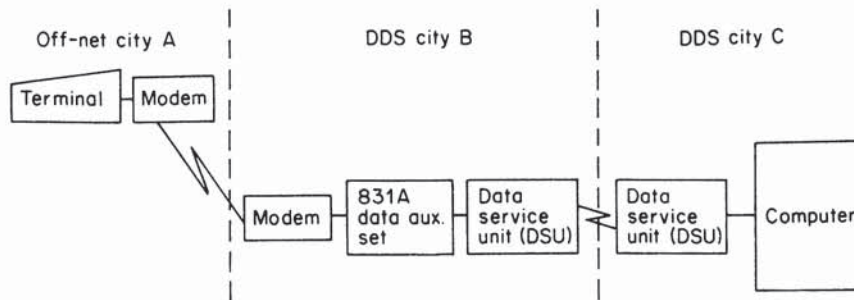
**Figure 4.89** Analog extension to DDS. In order to obtain an analog extension to a digital network, a device known as a data auxiliary set, which provides an interface between a modem and a service unit, must be installed

external clock, a second elastic store will be required. Figure 4.89 illustrates a typical analog extension to a DDS servicing area.

## Applications

As discussed in previous sections, the requirement for modems in an off-net analog extension could negate any real savings, gained in utilizing DDS. A network arrangement in the form of Figure 4.90, on the other hand, could easily achieve the high-performance characteristics inherent in DDS while reducing the overall costs of creating two independent data links.
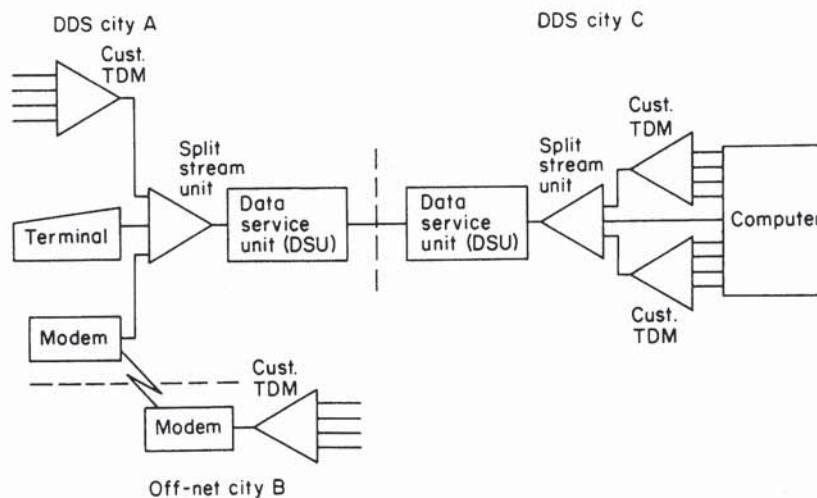


**Figure 4.90** Multiplexing over DDS utilizing split stream units. An inexpensive split stream unit, or limited-function synchronous multiplexer, can offer considerable flexibility when interfacing into the DDS network

**Table 4.39**  Split stream unit modes of operation

| DDS rate | Mode | SSU | Channel | Data | Rate |
|---|---|---|---|---|---|
| | 1 | 1 | 2 | 3 | 4 |
| 9600 | 1 | 9600 | | | |
| | 2 | 2400 | 7200 | | |
| | 3 | 7200 | 2400 | | |
| | 4 | 4800 | 4800 | | |
| | 5 | 2400 | 4800 | 2400 | |
| | 6 | 4800 | 2400 | 2400 | |
| | 7 | 2400 | 2400 | 2400 | 2400 |
| 4800 | 1 | 4800 | | | |
| | 2 | 1200 | 3600 | | |
| | 3 | 3600 | 1200 | | |
| | 4 | 2400 | 2400 | | |
| | 5 | 1200 | 2400 | 1200 | |
| | 6 | 2400 | 1200 | 1200 | |
| | 7 | 1200 | 1200 | 1200 | 1200 |
| 2400 | 1 | 2400 | | | |
| | 2 | 600 | 1800 | | |
| | 3 | 1800 | 600 | | |
| | 4 | 1200 | 1200 | | |
| | 5 | 600 | 1200 | 600 | |
| | 6 | 1200 | 600 | 600 | |
| | 7 | 600 | 600 | 600 | 600 |

This example shows the use of a device marketed by AT&T which is called a split stream unit (SSU). The SSU is similar to the multiplexer incorporated into now obsolete Bell System 209 data sets, in that the user can select various combinations of data transfer rates on up to four individual channels, up to the maximum capability of the DDS line. The SSU plugs directly into the DSU, with operational settings at one-half or one-quarter the specified DDS data rate. The unit provides local loop and remote loop testing of each individual channel and very effectively lifts the four speed restrictions of DDS service. Table 4.39 lists the operational modes of SSUs designed for use on 2400, 4800, and 9600 bps DDS circuits.

Of course, you can also obtain a variety of multiplexers manufactured by independent communications vendors that can be used in place of SSUs. Since DDS is a synchronous network that operates at a series of fixed data rates, you can also employ multiplexers as a means to concentrate data for communications over a DDS line. You can also use multiplexers to convert asynchronous data into synchronous data stream that is compatible with the data format for which DSUs are designed.

## KiloStream service

Although British Telecom's KiloStream service is similar to DDS, there are several significant differences that warrant discussion.

```
┌─────────┐    ----------> Transmit          (T)  -----     ┌─────────┐
│    D    │    ----------< Receive           (R)  -----     │    N    │
│         │    ----------> Control           (C)  -----     │         │
│    T    │    ----------< Indication        (I)  -----     │    T    │
│         │    ----------< Element timing    (S)  -----     │         │
│    E    │    ----------  Signal ground     (G)  -----     │    U    │
│         │    ----------  DTE common return (Ga) -----     │         │
└─────────┘                                                 └─────────┘
```
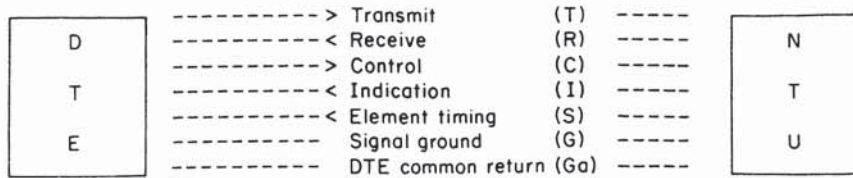
**Figure 4.91** ITU X.21 interface circuits

```
            -- 8 bit envelope ---

         ┌───┬───┬───┬───┬───┬───┬───┬───┐
         │ A │ I │ I │ I │ I │ I │ I │ S │
         └───┴───┴───┴───┴───┴───┴───┴───┘
```
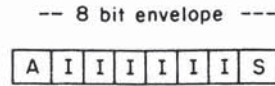
**Figure 4.92** KiloStream envelope encoding. (A Alignment bit which alternates between '1' and '0' in successive envelopes to indicate the start and stop of each 8-bit envelope. S Status bit which is set or reset by the control circuit and checked by the indicator circuit. I Information bits)

The British Telecom customer is provided with an interface device which is called a network terminating unit (NTU), which is similar to a DSU. The NTU provides an ITU interface for customer data at 2.4, 4.8, 9.6 or 48 kbps to include performing data control and supervision, which is known as structured data. At 64 kbps, the NTU provides an ITU interface for customer data without performing data control and supervision, which is known as unstructured data.

The NTU controls the interface via ITU recommendation X.21, which is the standard interface for synchronous operation on public data networks. An optional V.24 interface is available at 2.4, 4.8 and 9.6 bps while an optional V.35 interface can be obtained at 48 kbps. The X.21 interface is illustrated in Figure 4.91. Here the control circuit (C) indicates the status of the transmitted information—data or signaling, while the indication circuit (I) signals the status of information received from the line. The control and indication circuits control or check the status bit of an 8-bit envelope used to frame six information bits.

Customer data is placed into a 6 + 2 format to provide the signaling and control information required. This is known as envelope encoding and is illustrated in Figure 4.92.

The NTU performs signal conversion, changing unipolar non-return to zero signals from the V.21 interface into a diphase WAL 2 encoding format. This ensures that there is no dc content in the signal transmitted to the line, provides isolation of the electronic circuitry from the line, and provides transitions in the line signal to enable timing to be recovered at the distant end. Table 4.40 lists the NTU operational characteristics of KiloStream.

## The KiloStream network

In the KiloStream network, the NTUs on a customer's premises are routed via a digital local line to a multiplexer operating at 2.048 Mbps. This data rate is the European equivalent of the T1 line in the United States that operates at 1.544

**Table 4.40** KiloStream NTU operational characteristics

| Customer data rate (kbps) | DTE/NTU interface | Line data rate (kbps) | NTU operation |
|---|---|---|---|
| 2.4 | X.21 | 12.8 | 6+2 envelope encoding |
| 4.8 | X.21 | 12.8 | 6+2 envelope encoding |
| 9.6 | X.21 | 12.8 | 6+2 envelope encoding |
| 48 | X.21 | 64 | 6+2 envelope encoding |
| 64 | X.21 | 64 | No envelope encoding |
| 48 | X.21 bis/V.35 | 64 | 6+2 envelope encoding |
| 2.4 | X.21/V.24 | 12.8 | 6+2 envelope encoding |
| 4.8 | X.21/V.24 | 12.8 | 6+2 envelope encoding |
| 9.6 | X.21/V.24 | 12.8 | 6+2 envelope encoding |
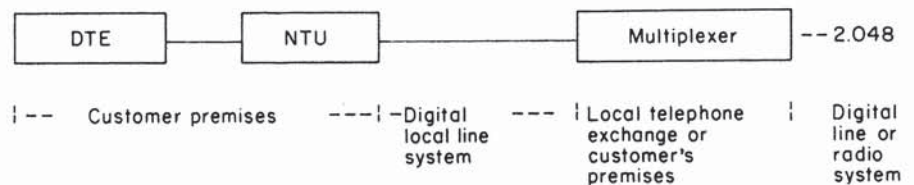


**Figure 4.93** The KiloStream connection

Mbps. The multiplexer can support up to 31 data sources and may be located at the local telephone exchange or on the customer's premises if traffic justifies. It is connected via a digital line or a radio system into the British Telecom KiloStream network as illustrated in Figure 4.93.

Unlike true CEPT, the 2.048 Mbps T-carrier used for KiloStream uses 31 DS0 channels. Normal, CEPT uses one channel for synchronization (framing) and a second channel for signaling. Since there is no direct voice signaling on KiloStream, time slot 16, which normally would carry that information, can be used for data. From an examination of Figure 4.94, the reader will note that a DTE operating rate of 2.4, 4.8 or 9.6 kbps results in a line rate of 12.8 kbps, which is precisely one-fifth of the DS0 64 kbps data rate.

## 4.11 CHANNEL SERVICE UNITS

A channel service unit (CSU) is a communications device that was developed to terminate a T1 line in North America and an E1 line in Europe. Since a data service unit (DSU) commonly includes a built-in CSU, it is natural for many readers to be confused with respect to the use and operating characteristics of CSUs. To alleviate this potential confusion, let us distinguish between modern standalone CSUs and CSUs built into DSUs.
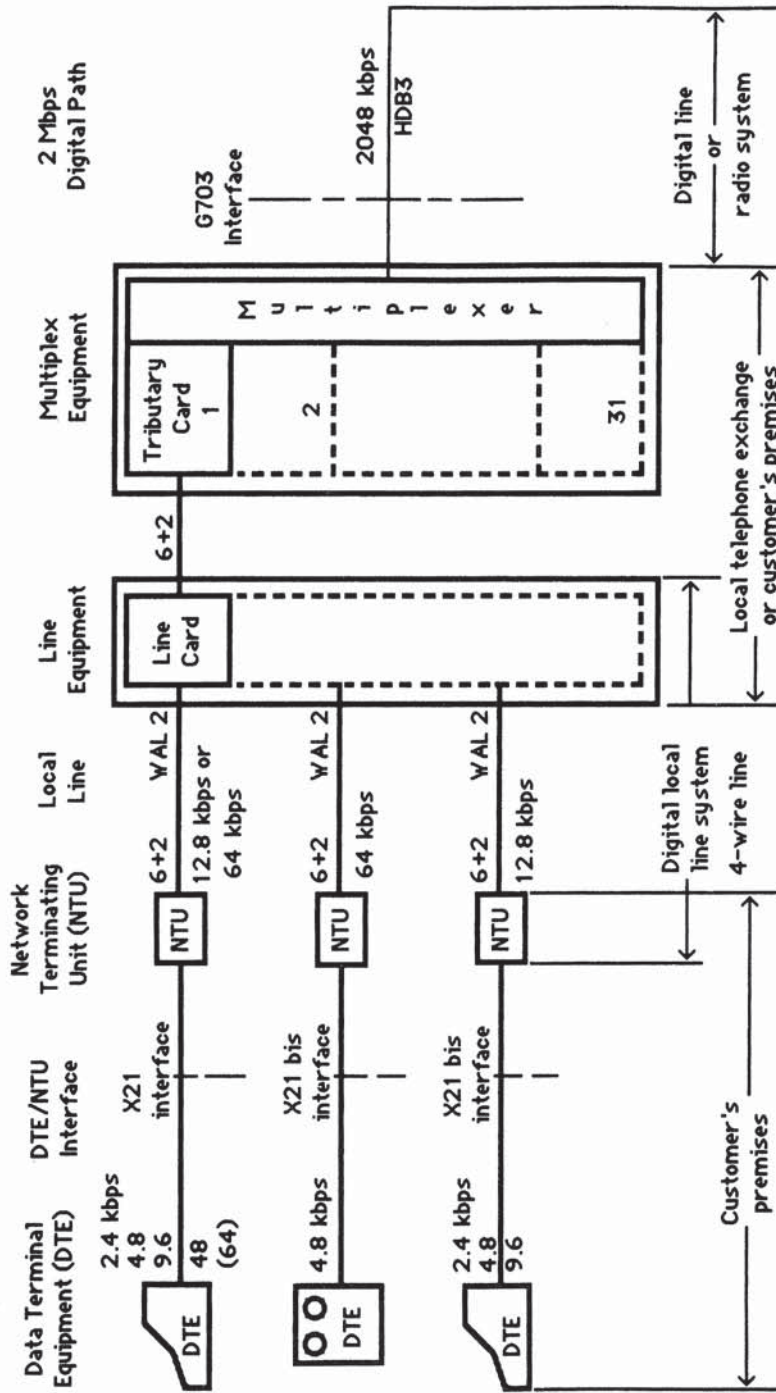
Figure 4.94  KiloStream structure

## Comparison to DSU/CSU

Modern CSUs designed for operation on T1 or E1 transmission facilities perform all of the functions previously described in Section 4.10 for the CSU portion of a DSU. In addition, a standalone CSU will frame data for transmission on T1/E1 facilities and suppress strings of binary zeros, which, if not suppressed, could result in the loss of timing by repeaters on a T1/E1 local loop between a subscriber and a carrier's central office serving the subscriber.

Today, most T1/E1 multiplexers, routers and other devices that operate at 1.544 Mbps or 2.048 Mbps include an option for a built-in DSU. A standalone CSU is thus required to perform line termination functions as well as framing and zero suppression. Figure 4.95 illustrates the use of CSUs in a T1 network segment.

To understand how CSUs operate, we will examine the framing structures used on North American and European T-carrier facilities, as well as several methods commonly used to suppress strings of zeros. Here we will use the term 'T-carrier' to collectively refer to a T1 line operating at 1.544 Mbps in North America and an E1 line operating at 2.048 Mbps in Europe.
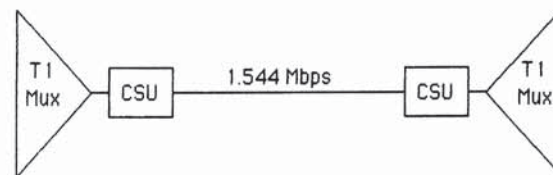


**Figure 4.95** Using CSUs on a T1 circuit. A standalone CSU used on a T1 circuit terminates the circuit as well as frames data, performs zero suppression, and recognizes and generates alarms

## North American framing

In North America, two framing formats are used on T1 lines: D4 and ESF.

### D4 framing

In North America the T1 signal represents a composite of 24 separate DS0 channels, each representing one PCM encoded voice signal digitized at 64 kbps. The 24 channels in a T1 signal are multiplexed in a round-robin order to ensure each channel is transmitted in turn and that every channel receives a turn prior to any channel receiving a second turn. To denote the beginning of each sequence of 24 digitized DS0 channels a special bit called the frame bit is prefixed to the beginning of each multiplexing cycle. Since each DS0 channel is encoded into a PCM word using eight data bits, 24 DS0 channels represents a sequence of 192 data bits. The full pattern of one frame bit and 192 data bits is called the DS1 (digital signal level 1) frame and represents a total of 193 bits. Since sampling occurs 8000

**A**

| F1 | 192D | F2 | 192D | • • • | F11 | 192D | F12 | 192D | F1 | 192D |
|---|---|---|---|---|---|---|---|---|---|---|

1 Frame = 1 frame bit followed by 192 data bits

Superframe = frames F1 through F12

**B**

| F1 | F2 | F3 | F4 | F5 | F6 | F7 | F8 | F9 | F10 | F11 | F12 |
|---|---|---|---|---|---|---|---|---|---|---|---|
| 1 | 0 | 0 | 0 | 1 | 1 | 0 | 1 | 1 | 1 | 0 | 0 |

Terminal Framing Ft    101010
Signal Framing Fs    001110
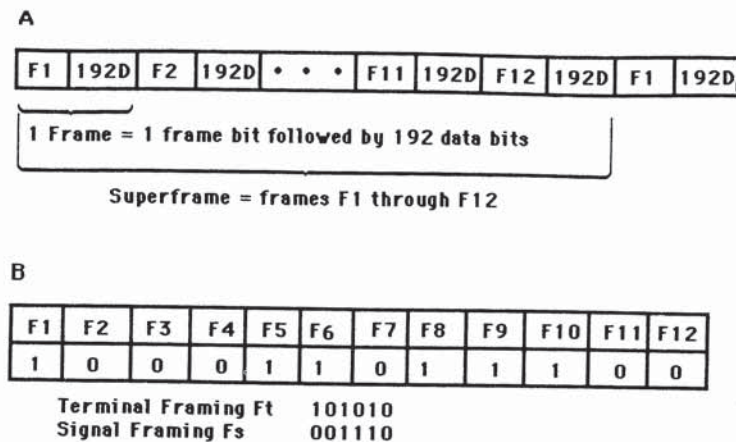
**Figure 4.96** D4 framing structure and framing pattern. The D4 framing pattern represents the hex characters 8CD which are continuously repeated. (a) D4 Framing Structure, (b) D4 Framing Pattern

times per second, 193 bits × 8000 samples per second results in the 1.544 Mbps operating rate of a T1 circuit. As this operating rate include 8000 frame bits, only the remaining 1.536 Mbps is actually available to the user.

One of the most popular methods of framing the DS1 signal is called D4 framing. This framing technique takes its name from the AT&T D4 channel bank used in that communications carrier's network. Under D4 framing the frame bits in 12 consecutive frames are grouped together to form a superframe whose frame bits are used to form a repeating pattern. Figure 4.96 illustrates the D4 framing structure and framing pattern. Under D4 framing, the 1.544 Mbps data stream must meet the following requirements.

(i) It must be encoded as a bipolar, AMI, non-return to zero signal to ensure that the signal has no dc component and can be transformer coupled, permitting the circuit to carry power for the repeaters.

(ii) Each pulse must have a 50% duty cycle with a nominal voltage of 3.0 V.

(iii) There can be no more than 15 consecutive '0's present in the data stream, which defines the minimum 1's density of the circuit.

(iv) The D4 framing pattern is embedded in the data stream.

The framing bits in the D4 superframe consist of six $F_t$ (terminal framing) bits that are used to synchronize the bit stream and six $F_s$ (signal framing) bits that are used to define multiframe boundaries.

The $F_t$ bit conveys a pattern of alternating 0's and 1's (101010), which is used to define frame boundaries, enabling one slot to be distinguished from another. Due to this, it is also known as a frame alignment signal. The $F_s$ bit conveys a pattern of 001110, which is used to define multiframe boundaries. This enables one frame to be distinguished from another, permitting frames 6 and 12 to be identified for the extraction of their signaling bits. Note that the composite D4 framing pattern represents the hex characters 8CD which are continuously repeated.

## Extended superframe format

A key limitation of the D4 format is the fact that obtaining a communications capability between devices on a T1 circuit for testing or control purposes required the use of a DS0 time slot. To alleviate this problem, as well as to provide the T1 user with additional capability, AT&T introduced an extended superframe format in early 1985. Although this new framing format require the installation of equipment that supports the frame format and has consequently been slowly introduced, by the late 1990s a majority of T1 circuits in North America conformed to it. Eventually, the extended superframe format can be expected to replace D4 framing.

Denoted as $F_e$ and ESF, the extended superframe format extends D4 framing to 24 consecutive frame bits—F1 to F24 as illustrated in Figure 4.97.

Unlike D4 framing in which the 12 framing bits form a specific repeating pattern, the ESF pattern can vary. ESF consists of three types of frame bits.

| F1 | 192D | F2 | 192D | • • • | F24 | 192D | F1 | 192D | • • |

1 Frame = 1 frame bit + 192 data bits

1 Extended Superframe = frames F1 through F24
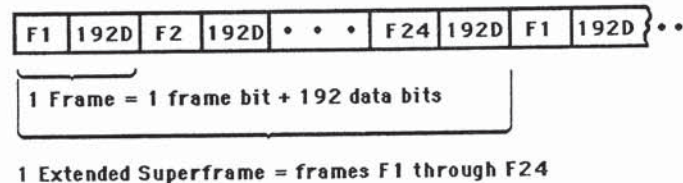
**Figure 4.97** The extended superframe

### Derived data link

The ESF 'd' bits, which represent a derived data link, are used by the telephone company to perform such functions as network monitoring to include error performance, alarm generation and reconfiguration to be passed over a T1 link. The 'd' bits appear in the odd frame positions, e.g., 1, 3, ..., 21, 23. Since they are used by 12 of the 24 framing bits, the 'd' bits represent a 4 kbps data link.

The data link formed by the 12 'd' frame bits is coded in to higher level data link control (HDLC) protocol format known as BX.25. Figure 4.98 illustrates the data link format carried by the 'd' bits.

One of the primary goals in the development of the BX.25 protocol was to provide a mechanism to extract performance information from ESF compatible CSUs. Doing so allows circuit quality monitoring without taking the circuit out of service and is a major advantage of ESF over D4 framing. Standard maintenance messages that are defined in AT&T's publication 54016 include messages that can return performance data concerning the number of errored seconds (ES), severely errored seconds (SES), and failed seconds (FS). An errored second is a second that contains one or more bit errors, while a severely errored second is considered to be a second with 320 or more bits in error. If ten consecutive severely errored seconds occur, this condition is considered as a failed signal state. Then, each signal in a failed signal state is considered to be a failed second. Table 4.41 summarizes the standard maintenance messages transmitted on the ESF data link.
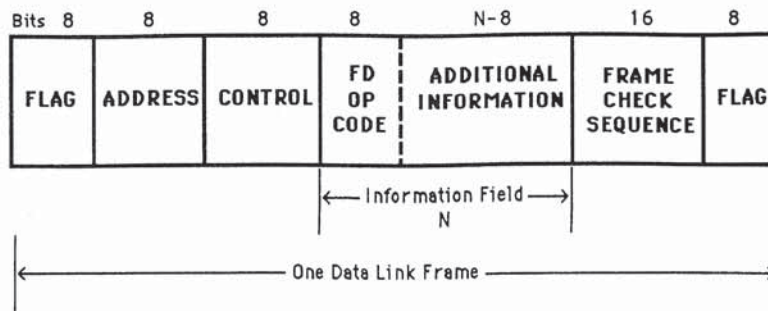
**Figure 4.98**  ESF data link format

**Table 4.41**  ESF data link maintenance messages

*Send One Hour Performance Data*
Upon receiving this command the ESF CSU will supply the following:
current status, elapsed time of current interval, ES and FS in the current 15-minute
interval, number of valid intervals, count of ES and FS in 24-hour register, ES and FS
during the previous four 15-minute intervals.

*Send 24-Hour 'ES' Performance Data*
Upon receiving this command, the ESF CSU will supply the following:
current status, elapsed time of current interval, ESs and FSs in current 15-minute
interval, number of valid intervals, count of ESs and FSs in 24-hour register, and ESs
during previous 96 15-minute intervals.

*Send 24-Hour 'FS' Performance Data*
Upon receiving this command, the ESF CSU will supply the following:
current status, elapsed time of current interval, ESs and FSs in the current 15-minute
interval, number of valid intervals, count of ESs and FSs in 24-hour register, and FSs
during previous 96 15-minute intervals.

*Reset Performance Monitoring Counters*
Upon receiving this command, the ESF CSU will reset all interval times and ES and
FS registers and supply the current status.

*Send Errored ESF Data*
Upon receiving this command, the ESF CSU will supply current data present in ESF
error event registers. Each count represents one error event (65535 max.).

*Reset ESF Register*
Upon receiving this command, the ESF CSU will reset the ESF error event register and
supply the current status.

*Maintenance Loop-Back (DLB)*
Energizes upon receiving the proper code embedded in the 4 kbps data link. This loop-back
loops through the entire CSU.

### Error check link

Frame bits 2, 6, 10, 14, 18 and 22 are used for a CRC-6 code. The 6-bit cyclic
redundancy check sum is used by the receiving equipment to measure the circuit's
bit error rate and represents 2 kbps of the 8 kbps framing rate. The CRC-6 code
yields an accuracy of 98.4% and the occurrence of a mismatch between the locally

generated check bit sequence and the received check bit sequence indicates that one or more bits in the extended superframe are in error.

To conform with ESF CRC-6 coding and reporting requirements, the use of an ESF compatible channel service unit is required. This CSU not only generates the CRC-6 but must also be capable of detecting CRC errors and storing a CRC error count over a 24-hour period. ESF compatible CSUs contain buffer storage which enables the device to store current line status information, including all error events and errored and failed seconds for the current 15-minute period and the previous 96 15-minute periods that represent the prior 24-hour period. To enable the carrier to retrieve this data, as well as to reset any or all counters and activate or deactivate loop-back testing on the local span line, the CSU must also have the ability to respond to network commands. An ESF compatible CSU must thus have the capability to send and receive data based upon the BX.25 formation via the 4 kbps data link.

A competing standard from the T1E1 committee requires the CSU to broadcast this information all of the time instead of upon request from the carrier. This standard is designed to overcome the potential problem resulting from an Interexchange Carrier (IEC) resetting CSU registers prior to a Local Exchange Carrier (LEC) reading the CSU registers.

The previously mentioned competing standard is officially known as ANSI T1.403. Under the T1.403 standard performance reports are transmitted every second and only three seconds of historical data are maintained. A T1.403 compliant CSU transmits a performance report of 112 bits about the current second and three previous seconds every second. This report contains a range of CRC-6 errors, frame bit errors, bipolar violations and other error event information for each second.

The key difference between AT&T Publication 54016 and the ANSI T1.403 ESF method is in the philosophy behind each ESF method since both use the 4 kbps data link for transmitting performance reports. AT&T's philosophy is one of in-line query—response under which parameters are organized into performance

**Table 4.42** ESF framing pattern

| Frame bit | Composition | Frame bit | Composition |
|-----------|-------------|-----------|-------------|
| 1 | d | 13 | d |
| 2 | C1 | 14 | C4 |
| 3 | d | 15 | d |
| 4 | 0 | 16 | 0 |
| 5 | d | 17 | d |
| 6 | C2 | 18 | C5 |
| 7 | d | 19 | d |
| 8 | 0 | 20 | 1 |
| 9 | d | 21 | d |
| 10 | C3 | 22 | C6 |
| 11 | d | 23 | d |
| 12 | 1 | 24 | 1 |

d = data link
C$x$ = CRC-6 bit $x$

registers for 15-minute intervals, and can be stored and retrieved for up to a 24-hour period. In comparison, under ANSI T1.403, parameters are broadcast every second and store only three seconds of data. Although AT&T and several communications carriers support both methods, AT&T appears to be migrating towards the ANSI method even though that method does not maintain historical data, nor does it allow user access to registers.

### Framing pattern

The third type of frame bits are used to generate the framing pattern. Here, frame bits 4, 8, 12, 16, 20 and 24 are used to generate the $F_e$ framing pattern whose composition is 001011. These six bits result in a 2 kbps framing pattern. In Table 4.42 you will find a summary of the ESF framing pattern.

## CEPT PCM-30 format

CEPT PCM-30 is a PCM format used for time division multiplexing of 30 voice or data circuits onto a single twisted pair cable using digital repeaters and is primarily used in Europe.

The standard CEPT frame is 32 channels × 8 bits/channel or 256 bits. With 8000 samples per second, the CEPT data rate becomes 8000 × 256, or 2.048 Mbps. Note that under this format there are no framing bits added to a frame as done under the North American T1 format. This is because the framing bits are carried within specific time slots. Thus, a European CSU must examine bits within a specific time slot.

### *Frame composition*

Each CEPT PCM-30 frame consists of 32 time slots to include 30 voice, one alignment and one signaling, with each time slot represented by eight bits. Since each PCM channel is sampled 8000 times per second, the standard CEPT-30 data rate is 32 × 8 × 8000, or 2.048 Mbps.

### Alignment signal

An alignment signal (0011011) is transmitted in bit positions 2 to 8 of time slot 0 in alternating frames. This signal is used to enable each channel to be distinguished at the receiver. Bit position 1 in time slot zero carries the International bit, while frames not containing the frame alignment signal are used to carry National and International signaling and alarm indication for loss of frame alignment. Figure 4.99 illustrates the composition of the CEPT-30 frame and multiframe, where the multiframe consists of 16 frames, numbered from frame 0 to frame 15.

To avoid imitation of the frame alignment signal, alternating frames fix bit 2 to a 1 in time slot zero which is the reason why a 1 is entered into that bit position for odd time slot 0 frames.

**16 Frames = 1 Multiframe**

| 0 | 1 | 2 | 3 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 | 13 | 14 | 15 |

**32 time slots per frame**

**Time Slots**

| 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 | 13 | 14 | 15 | 16 | 17 | 18 | 19 | 20 | 21 | 22 | 23 | 24 | .... | 31 |

**Frame Alignment Signal**                    **Multiframe Alignment Signal**

| I | 0 | 0 | 1 | 1 | 0 | 1 | 1 | even frames          | 0 | 0 | 0 | 0 | X | Y | X | X | frame 0

**FAS**                                    **MAS**

| I | 1 | A | N | N | N | N | N | odd frames          | A | B | C | D | A | B | C | D | frames 1–15
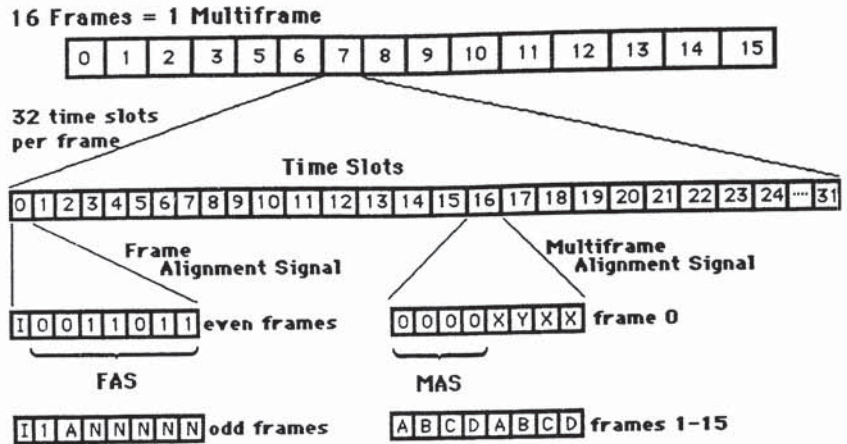
**Figure 4.99** CEPT PCM-30 frame and multiframe composition (i international bit, N national bit, A alarm indication signal, FAS frame alignment signal, ABCD ABCD signaling bits, X extra bit for signaling, Y loss of multiframe alignment, MAS multiframe alignment signal)

## CEPT CRC option

For enhanced error monitoring capability, CEPT PCM-30 includes a CRC-4 option. Under this option, a group of eight frames known as a submultiframe (SMF) is treated as a long binary number. This number is multiplied by $X^4$ (10000) and divided by $X^4 + X + 1$ (10011). The four-bit remainder is transmitted in bit position 1 (1 bit in Figure 4.99) in time slot zero in even frames which contain the frame alignment signal. After the receiver computes its own CRC-4 check it uses bit position 1 in time slot zero of frames 13 and 15 for CRC error performance reporting. Table 4.43 summarizes how these bits are used for CRC error performance reporting purposes.

**Table 4.43** CEPT PCM-30 CRC error performance reporting

| Bit 1 Frame 13 | Bit 1 Frame 15 | |
| --- | --- | --- |
| 1 | 1 | CRC for SMF I, II error free |
| 1 | 0 | SMF II in error, SMF I error free |
| 0 | 1 | SMF II error free, SMF I in error |
| 0 | 0 | Both SMF I and II in error |

## T-carrier signal characteristics

As previously discussed, there are several advantages to transmitting T-carrier signals in a bipolar alternate mark inversion (AMI) format, including the absence of

a dc component to the signal and the ability to obtain clock recovery from the signal in an all-ones condition. Unfortunately, the disadvantage associated with this signaling method is that a sequence of spaces is encoded as a period of zero voltage or no signal and repeaters on a span line cannot recover clocking without a signal occurring every so often.

For repeaters to properly recover clocking, a certain number of binary ones must be contained within a transmitted signal. The process used to ensure a minimum number of ones flows on a T1 or E1 line is called zero suppression and is performed by CSUs. Since North American and European approaches to this problem differ, we will examine the encoding methods used to ensure an appropriate signal contains a minimum number of binary ones for each location.

## North America

In North America, AT&T publication 62411 sets the ones density requirement to be 'n' ones in each window of $8 \times (n+1)$ bits, where $n$ varies from 1 to 23. This means that a T1 carrier cannot have more than 15 consecutive zeros ($n = 1$) and there must be approximately three ones in every 24 consecutive bits ($n = 2$ to 23). Two methods used to provide this minimum ones density include binary 7 zero code suppression and binary 8 zero substitution.

### Binary 7 zero code suppression

Under the binary 7 zero code suppression method a binary one is substituted in bit position 7 of each time slot if all eight positions are zeros. An example of this method of ensuring a minimum ones density is illustrated at the top of Figure 4.100(a). Although it might appear wiser to select the least significant bit for inversion, this cannot be done since the setting of a frame bit to zero when bit positions 2 to 8 in the previous time slot were set to zero would result in a string of 16 consecutive zeros if the bits in the time slot following the frame bit were zero and bit position 8 was used for substitution. Figure 4.100(b) illustrates this worst-case scenario which explains why bit position 7 in each time slot is used for bit value inversion to ensure a minimum ones density.

If a data channel contains all 0s, the data can be corrupted due to B7 zero suppression. Therefore a data channel normally is restricted to seven usable data bits, with one bit in the data channel set to a 1. This prevents the data channel from being corrupted, but also limits its data rates to 56 kbps.

When one bit is set to a 1 on a DS0 channel, the channel is known as a non-clear channel. The 56 kbps on a non-clear channel is also known as a DS-A channel.

A T1 clear channel is one in which all 64 kbps in each DS0 are usable. On private microwave systems, B7 zero code suppression is normally not required, permitting clear channel capability.

### Binary 8 zero substitution

The binary 8 zero substitution (B8ZS) technique was developed by Bell Laboratories and is now sanctioned by the ITU for use in North America. This

**A**

**If this byte composition occurs**

| bit 1 | bit 2 | bit 3 | bit 4 | bit 5 | bit 6 | bit 7 | bit 8 |
|-------|-------|-------|-------|-------|-------|-------|-------|
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |

**Result of B7 zero substitution**

| bit 1 | bit 2 | bit 3 | bit 4 | bit 5 | bit 6 | bit 7 | bit 8 |
|-------|-------|-------|-------|-------|-------|-------|-------|
| 0 | 0 | 0 | 0 | 0 | 0 | 1 | 0 |

**B**

**DS-0 Frame 24**

| bit 1 | bit 2 | bit 3 | bit 4 | bit 5 | bit 6 | bit 7 | bit 8 | frame N |
|-------|-------|-------|-------|-------|-------|-------|-------|---------|
| 1 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | |

**DS-0 Frame 1**

| bit 1 | bit 2 | bit 3 | bit 4 | bit 5 | bit 6 | bit 7 | bit 8 |
|-------|-------|-------|-------|-------|-------|-------|-------|
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 9 | 10 | 11 | 12 | 13 | 14 | 15 | 16 |

**Substitution bit to a "1"**
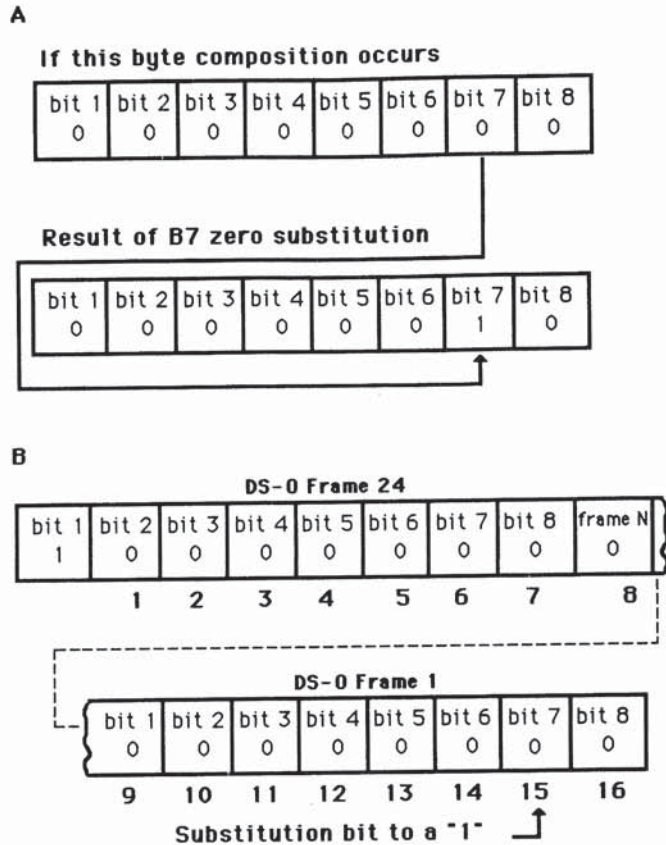
**Figure 4.100** B7 zero code suppression: (a) B7 zero code suppression example; (b) worst-case scenario

method of ensuring a minimum ones density was placed into operation during the mid 1980s and offers a significant improvement over binary 7 zero code suppression as it both maintains a minimum ones density and also provides a clear channel capability, permitting each DS0 channel to carry data at 64 kbps.

Under B8ZS coding, each eight consecutive 0s in a byte are removed and replaced by a B8ZS code. If the pulse preceding an all-zero byte is positive, the inserted code is $000 + - 0 - +$. If the pulse preceding an all-zero byte is negative, the inserted code is $000 - + 0 + -$. Figure 4.101 illustrates the use of B8ZS coding in which an all-zeros byte is replaced by one of two binary codes, with the actual code used based upon whether the pulse preceding the all-zeros byte was positive or negative.

Both examples result in bipolar violations occurring in the fourth and seventh bit positions. Both carrier and customer equipment must recognize these codes as legitimate signals and not as bipolar violations or errors for B8ZS to work to enable a receiver to recognize the code and restore the original eight zeros.
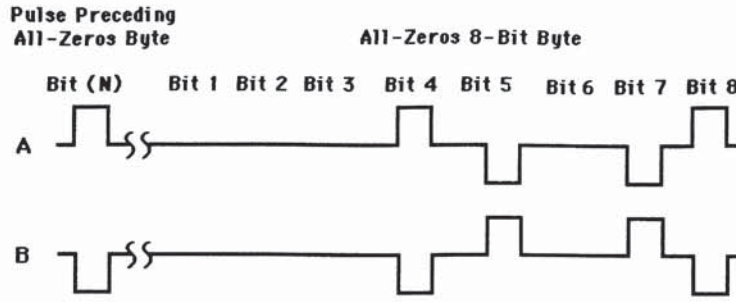
**Figure 4.101**   B8ZS coding

*Europe*

In Europe the high density bipolar 3-zero maximum (HDB3) coding is used by CEPT PCM-30 to obtain a minimum ones density for clock recovery from received data. Under HDB3, the data stream to be transmitted is monitored for any group of four consecutive zeros. A four-zero group is then replaced with an HDB3 code. Two different HDB3 codes are used to ensure that the bipolar violation pulses from adjacent four-zero groups are of opposite polarity as indicated in Figure 4.102. The selection of the HDB3 code is based upon whether there was an odd or even number of ones since the last bipolar violation (BV) occurred. If an odd number of ones occurred since the previous bipolar violation, the coding method in Figure 4.102 (a), is used to replace a sequence of four zeros. If an even number of ones occurred since the previous bipolar violation the coding method in Figure 4.102(b), is used to replace a sequence of four zeros.
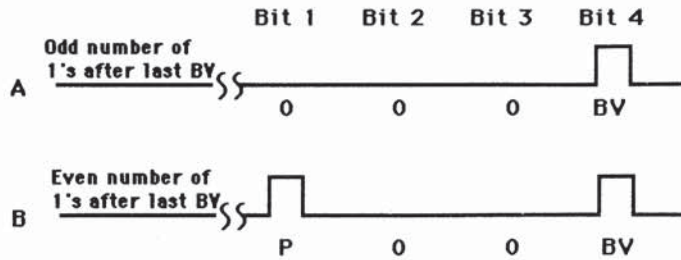


**Figure 4.102**   HDB3 coding (P polarity bit, BV bipolar violation)

## 4.12  PARALLEL INTERFACE EXTENDERS

When an application arises that will require batch processing at a site remotely located from a mainframe computer, several approaches can be considered to satisfy this requirement.

A traditional approach is the establishment of a remote batch processing operation. The establishment of this type of operation normally requires the procurement of several communications components in addition to the remote batch terminal. First, a communications controller must be installed and interfaced to the computer if such a device does not already exist at the computational facility. This controller, in conjunction with the computer, performs such tasks as character assembly and disassembly, transmission error checking by generating check characters from the received data blocks, and comparing the check character to the check character generated by the remote batch terminal, as well as performing numerous traffic management functions. Next, a teleprocessing software module will be added and integrated to the computer's operating system to perform and control the transmission discipline. This software may not only be costly but may affect computer performance since it typically requires hundreds of thousands to millions of or more memory locations. Last, a transmission medium and either a high-speed modem or DSU to translate the signals into an acceptable form for transmission must be installed. This traditional remote batch processing operation is illustrated in Figure 4.103.
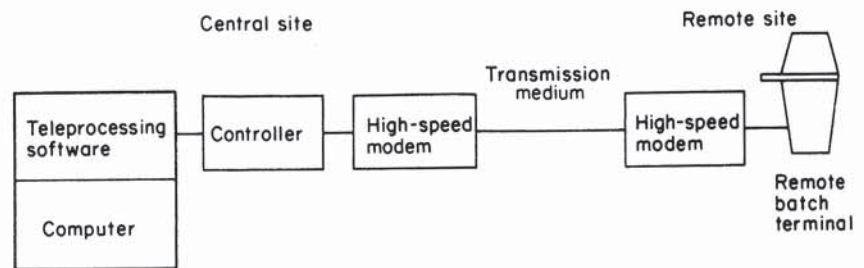


**Figure 4.103** Traditional remote batch processing. A large portion of the computer's memory may be reserved for teleprocessing software

Another problem which may arise is the compatibility of the remote batch terminal to the computer system already installed. If the terminal obtained does not support the protocol of the computer's teleprocessing software an emulator may be required to provide an acceptable transmission link. In any event, transmission from the computer to the terminal will most likely require code conversion, since most terminals cannot accept the computer's native code. Realizing these problems, a device was introduced which permits transmission from selected computers to a variety of computer peripherals without the necessity of the addition of special teleprocessing software or a communications controller. This device is called a parallel interface extender.

Another application that can require the extension of the parallel interface of a mainframe computer channel involves the connection of remote LANs and high-speed peripheral devices to a centrally located mainframe. Since the device in effect extends the parallel channel of the mainframe, another name for this device is a channel extender.

## Extender operation

A parallel interface or channel extender is a device which translates the parallel protocol of an input/output or channel (I/O) channel from such devices as a computer or selected computer peripheral units into a serial protocol which is suitable for transmission over a normal serial communications link. At the other end of the link, another parallel interface extender or similar operating device translates the serial protocol formed by the first parallel interface extender back into the original parallel protocol transmitted by the I/O channel for reception by devices similar to the standard peripherals used for local data processing at a computer center. From a broad viewpoint, a parallel interface extender can be compared as being similar in performance to a multiplexer, combining the data from a number of leads of the parallel I/O channel into a single bit stream for transmission over a serial communications link, as shown in Figure 4.104.
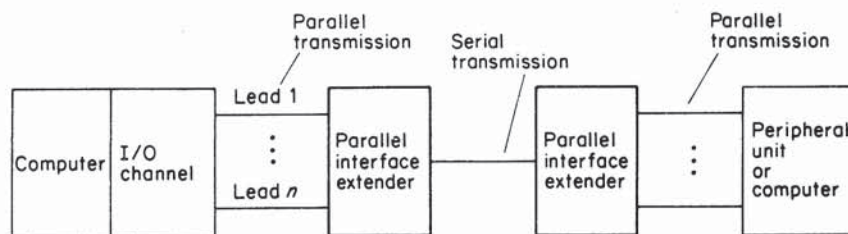


**Figure 4.104** Parallel interface or channel extender operation. Operation is similar to a multiplexer, combining the data from a number of leads from a computer or peripheral I/O channel into a single bit stream for transmission

By permitting a computer to utilize its regular I/O channel when communicating with a remotely located peripheral device, the necessity of obtaining specialized communications software is alleviated, and communications with remote peripheral units can then take place with the same software which is regularly used for communicating with the local peripherals at the computer center. Depending upon the computer configuration in use, a parallel interface extender may reduce operating system software requirements from 10 to 30% or more, when compared with communicating to remote devices by using a line controller and the required teleprocessing software modules. Another advantage obtained through the use of parallel interface extenders is the ability to program remote applications in Fortran, Cobol and other higher level languages, as well as in assembly language, using the READ, WRITE statements of Fortran and Cobol or the GET and PUT macros of assembly language to perform input and output remote peripheral functions. In addition to the translation of parallel to serial and serial to parallel data, to perform remote peripheral functions the device encodes commands and status information into a serial bit stream at the transmission end of the link while the device at the receiving end of the link performs a decoding function to reconstruct the original commands, status, and data before passing such information to the remote peripheral unit.

## Extender components

A parallel interface extender consists of a control unit and one or more line module groups, as shown in Figure 4.105. The control unit of the extender connects to the multiplexer channel of a computer and emulates the functions of several peripheral control units, such as card readers, card punches, magnetic tapes, and line printers, thus supporting the computer's standard software, as previously mentioned. Each line module group is connected to the control unit of the extender on one side and provides an interface for the connection of dedicated, switched, or leased lines on the opposite side. The line module contains all necessary line control and error control components, as well as a built-in modem which alleviates the necessity of having special communications software such as IBM's virtual telecommunications access method (VTAM) or synchronous data link control (SDLC), a communications controller, line adapter, and a separate modem. For alternate high-speed data transfer in either direction, half-duplex line modules can be used while full-duplex line modules provide data transfer in both directions at the same time. For data transfer over the switched network, half-duplex line modules are normally used while either half-duplex or full-duplex line modules can be used with dedicated or leased four-wire lines.
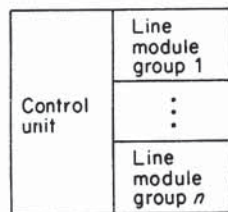
**Figure 4.105** Parallel interface extender components. A parallel interface extender consists of a control unit and one or more line module groups. In addition, some manufacturers provide modems or DSUs/CSUs built in to the device

## Application examples

One example of the use of a Parallel interface extender is a product that connects to an IBM system central processing unit via the system's byte multiplexer channel and can be used for such diverse applications as permitting two computers to communicate with each other in their native code, as in Figure 4.106; or it can be used to connect a computer to a variety of local or remotely located peripheral units,
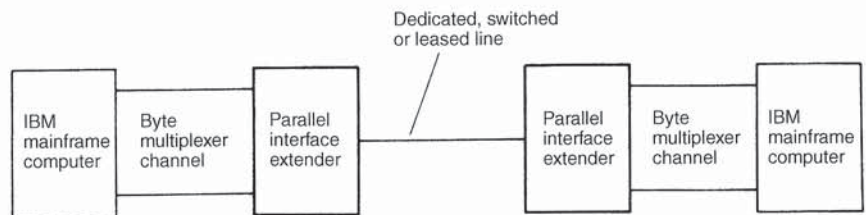
**Figure 4.106** Intercomputer communications using a parallel intenace extender. Using a parallel interface extender, the parallel transmission of the byte multiplexer channel of a computer is converted into a serial data stream for transmission
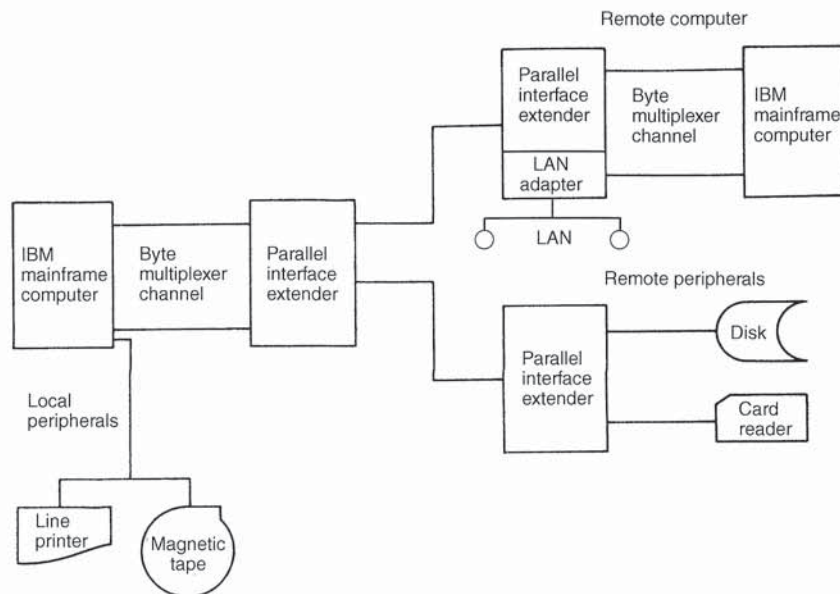
**Figure 4.107** Local and remote peripherals as well as computers can be serviced. Through the utilization of a parallel interface extender, remote peripherals as well as remote computers are referenced by the central computer as if they were local devices

as shown in Figure 4.107. In examining Figure 4.107, note the integration of a LAN adapter into one of the remote parallel interface extenders (channel extender). Some extenders support the addition of Token-Ring or Ethernet LAN modules, permitting remote LANs as well as remote peripherals and remote mainframes to be integrated into a common network infrastructure.

Supporting data transmission rates from 4800 bps to 2.048 mbps, the parallel interface extender permits a level of data transfer that can be matched to the operating speed of most peripheral devices. Parallel interface extenders permit peripherals to be transferred to remote locations for other applications than those originally required; hence, the use of such equipment warrants further examination.

# REVIEW QUESTIONS

**4.1.1** What is the difference between an acoustic coupler and a modem with respect to their line connection?

**4.1.2** What does the term 'Bell System' compatibility mean when discussing the operational characteristics of a modem?

**4.1.3** If an originate mode acoustic coupler transmits a mark at $f_1$ and a space at $f_2$ and receives a mark at $f_3$ and a space at $f_4$, what would be the corresponding frequencies of an answer mode coupler to ensure communications compatibility?

**4.1.4** Discuss the conventional utilization of originate and answer mode couplers and modems.

**4.1.5** Why is it more likely than not that an American using his or her portable personal computer in Europe would not be able to communicate with a computer located in Europe?

**4.2.1** What is a carrier signal? By itself, does it convey any information? Why?

**4.2.2** How can the characteristics of a carrier signal be altered?

**4.2.3** What is the difference between a bit per second and a baud? When can they be equivalent? When are they not equivalent?

**4.2.4** What is the Nyquist relationship and why does it require modem designers to develop multilevel phase-shift keying modulation schemes for modems to operate at high data rates?

**4.2.5** What does the signal constellation pattern of a modem represent? What is the normal relationship between the density of the signal constellation pattern and the susceptibility of a modem to transmission impairments?

**4.2.6** Discuss the difference between Trellis-coded modulation and conventional quadrature amplitude modulation with respect to the density of the signal constellation and the susceptibility of a modem employing each modulation technique to transmission impairments.

**4.2.7** How could you use a reverse channel? What is the difference between a reverse channel and a secondary channel?

**4.2.8** What is a multiport modem and under what circumstances should you consider using this device?

**4.2.9** Why are Bell System 212-type modems operating at 1200 bps sometimes compatible with V.22 modems while at other times they are incompatible?

**4.2.10** Why are Bell System 202 type modems incompatible with ITU V.23 modems?

**4.2.11** Discuss the compatibility of an ITU V.26 modem employing a pattern A phase change with a similar modem using the pattern B phase change.

**4.2.12** Explain why the V.29 signal constellation pattern forms a mirror image.

**4.2.13** What are the key differences between the ITU V.32 and V.32 bis modems with respect to their operating rates and fallback/fall-forward capability?

**4.2.14** Does the V.32 terbo represent an industry standard modem?

**4.2.15** Why do some V.34 modems have a maximum operating rate of 28.8 kbps while other V.34 modems can operate at 33.6 kbps?

**4.2.16** What is the maximum operating rate of a 56 kbps modem when a transmission path flows over two analog–digital conversion points?

**4.2.17** Assume two modems connected to a leased line have both local and digital loop-back self-testing features. If communications were disabled, discuss the tests you would perform to determine if the line or one or both modems caused the communications failure.

**4.2.18** What is a wraparound unit? When would it be used?

**4.3.1** What are the three functions primarily associated with intelligent modems?

**4.3.2** Why can the term 'Hayes compatible' be misleading?

**4.3.3** What are result codes?

**4.3.4** What command line entry would you use to set the carrier wait time of an intelligent modem to 120 s?

**4.3.5** How does an intelligent modem correct detected errors?

**4.3.6** What is the purpose of flow control?

**4.3.7** Discuss three methods that are used for flow control.

**4.3.8** At what level does the MNP protocol perform error detection and correction?

**4.3.9** What is the difference between V.42 and V.42 bis?

**4.3.10** How does run-length encoding function under MNP class 5?

**4.3.11** What is the major problem affecting transmission on a cellular facility?

**4.4.1** What are the advantages and disadvantages in using multiport modems instead of separate modems and multiplexers?

**4.4.2** What function does a DCE option perform? How would you obtain a DCE option through the use of a cable?

**4.5.1** What is the key difference between multipoint modems and conventional modems? Discuss the differences in throughput obtained on a multipoint circuit as the average block size transmitted increases.

**4.5.2** Why is the effect of multipoint modems on satellite circuits minimal?

**4.6.1** How does a security modem operate?

**4.6.2** What two data elements would a network administrator enter into a security modem?

**4.6.3** Discuss two limitations associated with the use of security modems.

**4.7.1** Discuss the relationship between the data rate, wire gauge, and transmission distance.

**4.7.2** What is the difference between a line driver and a limited-distance modem?

**4.8.1** Discuss the difference between line drivers and limited-distance modems with respect to matching components.

**4.8.2** If you are using 26 AWG cable, what resistance would your transmission encounter if the cable distance was 2000 feet?

**4.8.3** What is the primary benefit from the use of a multiport limited distance modem?

**4.8.4** Assume your organization has eight terminals that require a connection to the computer system located in the same building 600 feet distant. If the terminals operate at 2400 bps and can only transmit pulses 400 feet before the pulses become distorted, determine the most economic method to connect the terminals given the following cost for equipment and cable:

| | |
|---|---|
| cable cost per foot | $ 0.50 |
| line driver | $100.00 |
| multiport limited-distance modem (4 port) | $250.00 |

**4.9.1** Describe two limitations that can restrict transmission on a subscriber line to 33.6 kbps.

**4.9.2** Why are many cable TV systems limited to a unidirectional transmission capability?

**4.9.3** Why is the actual transmission obtained on a 6 MHz cable modem channel significantly less than the capacity of the channel?

**4.9.4** What is the purpose of using 2B1Q coding on an HDLS transmission facility?

**4.9.5** What is the purpose of using an asymmetrical transmission method that employs a small band of frequencies for transmission from a subscriber to a central office?

**4.10.1** Why is the probability of a transmission error occurring on a digital transmission facility less than that on an analog transmission facility?

**4.10.2** Why do digital transmission facilities employ bipolar signaling?

**4.10.3** Define a bipolar violation.

**4.10.4** Why is a string of six zeros converted into a zero suppression sequence on DDS?

**4.10.5** What is an intentional bipolar violation? How is it used on DDS?

**4.10.6** How can you transmit asynchronous data on DDS?

**4.11.1** What is the primary difference between a CSU designed to operate on a T1 or E1 circuit and the CSU of a DSU/CSU designed to operate on a DDS transmission facility?

**4.11.2** What is the actual data rate available for use on a T1 line?

**4.11.3** How many bits are included in a superframe?

**4.11.4** Discuss the advantages of the use of the extended superframe format over D4 framing.

**4.11.5** How many bits are in an extended superframe?

**4.11.6** Discuss the use of each of the three types of frame bits in the extended superframe.

**4.11.7** How many bits are in a CEPT frame? Where are the framing bits in a CEPT frame?

**4.11.8** How many bits are contained in the CEPT multiframe?

**4.11.9** What is the advantage of binary 8 zero substitution over binary 7 zero code suppression?

**4.12.1** What is the advantage of using a parallel interface extender with respect to teleprocessing software?

**4.12.2** How could you use a central computer facility to distribute reports to several remotely located printers connected to the central computer via parallel interface extenders?

10BASE-T. However, emerging gigabit Ethernet does not provide a 10-fold processing increase above Fast Ethernet, as the former uses several techniques that alter the minimum length of frames and transmits a sequence of frames from a workstation (referred to as packet bursting) to extend its transmission distance. Thus, there is no direct correlation between the frame processing capability of gigabit Ethernet and Fast Ethernet.

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