ENSEMBLE MODEM STRUCTURE FOR IMPERFECT TRANSMISSION MEDIA

BACKGROUND OF THE INVENTION

1. Field of the Invention:

The invention relates generally to the field of data communications and, more particularly, to a high speed modem.

2. Description of the Prior Art:

Recently, specially designed telephone lines for the direct transmission of digital data have been introduced. However, the vast majority of telephone lines are designed to carry analog voice frequency (VF) signals. Modems are utilized to modulate VF carrier signals to encode digital information on the VF carrier signals and to demodulate the signals to decode the digital information carried by the signal.

Existing VF telephone lines have several limitations that degrade the performance of modems and limit the rate at which data can be transmitted below desired error rates. These limitations include the presence of frequency dependent noise on the VF telephone lines, a frequency dependent phase delay induced by the VF telephone lines, and frequency dependent signal loss.

Generally, the usable band of a VF telephone line is from slightly above zero to about four kHz. The power spectrum of the line noise is not uniformly distributed over frequency and is generally not determinative. Thus, there is no *a priori* method for determining the distribution of the noise spectrum over the usable bandwidth of the VF line.

Additionally, a frequency-dependent propagation delay is induced by the VF telephone line. Thus, for a complex multi-frequency signal, a phase delay between the various components of the signal will be induced by the VF telephone line. Again, this phase delay is not determinative and must be measured for an individual VF telephone line at the specific time that transmission takes place.

Further, the signal loss over the VF telephone line varies with frequency. The equivalent noise is the noise spectrum component added to the signal loss component for each carrier frequency, where both components are measured in decibels (dB).

Generally, prior art modems compensate for equivalent line noise and signal loss by gear-shifting the data rate down to achieve a satisfactory error rate. For example, in U.S. patent 4,438,511, by Baran, a high speed modem designated SM9600 Super Modem manufactured by Gandalf Data, Inc., is described. In the presence of noise impairment, the SM9600 will "gear shift" or drop back its transmitted data rate to 4800 bps or 2400 bps. The system described in the Baran patent transmits data over 64 orthogonally modulated carriers. The Baran system compensates for the frequency dependent nature of the noise on the VF line by terminating transmission on carriers having the same frequency as the frequency of large noise components on the line. Thus, Baran gracefully degrades its throughput by ceasing to transmit on carrier frequencies at the highest points of the VF line noise spectrum. The Baran system essentially makes a go/no go decision for each carrier signal, depending on the distribution of the VF line noise spectrum. This application reflects a continuation of the effort initiated by Baran.

Most prior art systems compensate for frequency dependent phase delay induced by the VF line by an equalization system. The largest phase delay is induced in frequency components near the edges of the usable band. Accordingly, the frequency components near the center of the band are delayed to allow the frequency components at the outside of the band to catch up. Equalization generally requires additional circuitry to accomplish the above-described delays.

A further problem associated with two way transmission over the VF telephone line is that interference between the outgoing and incoming signals is possible. Generally, separation and isolation between the two signals is achieved in one of three ways:

(a) Frequency multiplexing in which different frequencies are used for the different signals. This method is common in modem-based telecommunication systems.

(b) Time multiplexing, in which different time segments are used for the different signals. This method is often used in half-duplex systems in which a transmitter relinquishes a channel only after sending all the data it has. And,

(c) Code multiplexing, in which the signals are sent using orthogonal codes.

All of the above-described systems divide the space available according to constant proportions fixed during the initial system design. These constant proportions, however, may not be suitable to actual traffic load problem presented to each modem. For example, a clerk at a PC work station connected to a remote host computer may type ten or twenty characters and receive a full screen in return. In this case, constant proportions allocating the channel equally between the send and receive modems would greatly overallocate the channel to the PC work station clerk. Accordingly, a modem that allocates channel capacity according to the needs of the actual traffic load situation would greatly increase the efficient utilization of the channel capacity.

SUMMARY OF THE INVENTION

The present invention is a high-speed modem for use with dial-up VF telephone lines. The modem utilizes a multicarrier modulation scheme and variably allocates data and power to the various carriers to maximize the overall data transmission rate. The allocation of power among the carriers is subject to the constraint that the total power allocated must not exceed a specified limit.

In a preferred embodiment, the modem further includes a variable allocation system for sharing control of a communication link between two modems (A and B) according to actual user requirements.

Another aspect of the invention is a system for compensating for frequency dependent phase delay and preventing intersymbol interference that does not require an equalization network.

According to one aspect of the invention, quadrature amplitude modulation (QAM) is utilized to encode data elements of varying complexity on each carrier. The equivalent noise component at each carrier frequency is measured over a communication link between two modems (A and B).

As is known in the art, if the bit error rate (BER) is to be maintained below a specified level, then the power required to transmit a data element of a given complexity on a given carrier frequency must be increased if the equivalent noise component at that frequency increases. Equivalently, to increase data complexity, the signal to noise ratio, S/N, must be increased.

In one embodiment of the present invention, data and power are allocated to maximize the overall data rate within external BER and total available power constraints. The power allocation system computes the marginal required power to increase the symbol rate on each carrier from n to n + 1 information units. The system then allocates information units to the carrier that requires the least additional power to increase its symbol rate by one information unit. Because the marginal powers are dependent on the values of the equivalent noise spectrum of the particular established transmission link, the allocation of power and data is specifically tailored to compensate for noise over this particular link.

According to another aspect of the invention, a first section of the symbol on each carrier is retransmitted to form a guard-time waveform of duration $T_E + T_{PH}$ where T_E is the duration of the symbol and T_{PH} is the duration of the first section. The magnitude of T_{PH} is greater than or equal to the maximum estimated phase delay for any frequency component of the waveform. For example, if the symbol is represented by the time series, $x_0 \ldots x_{n-1}$, transmitted in time T_E; then the guardtime waveform is represented by the time series, $x_0 \ldots x_{n-1}, x_0 \ldots x_{m-1}$, transmitted in time $T_E + T_{PH}$. The ratio that m bears to n is equal to the ratio that T_{PH} bears to T_E.

At the receiving modem, the time of arrival, T_0, of the first frequency component of the guard-time waveform is determined. A sampling period, of duration T_E, is initiated a time $T_0 + T_{PH}$.

Accordingly, the entire symbol on each carrier frequency is sampled and intersymbol interference is eliminated.

According to a still further aspect of the invention, allocation of control to the transmission link between modems A and B is accomplished by setting limits to the number of packets that each modem may transmit during one transmission cycle. A packet of information comprises the data encoded on the ensemble of carriers comprising one waveform. Each modem is also constrained to transmit a minimum number of packets to maintain the communication link between the modems.

Thus, even if one modem has no data to transmit, the minimum packets maintain timing and other parameters are transmitted. On the other hand, if the volume of data for a modem is large, it is constrained to transmit only the maximum limited number of packets, N, before relinquishing control to the other modem.

In practice, if modem A has a small volume of data and modem B has a large volume of data, modem B will have control of the transmission link most of the time. If control is first allocated to modem A it will only transmit the minimal number, I, of packets. Thus A has control for only a short time. Control is then allocated to B which transmits N packets, where N may be very large. Control is again allocated to modem A which transmits I packets before returning control to B.

Thus, allocation of control is proportional to the ratio of I to N. If the transmission of the volume of data on modem A requires L packets, where L is between I and N, then the allocation is proportional to the ratio of L to N. Accordingly, allocation of the transmission link varies according to the actual needs of the user.

Additionally, the maximum number of packets, N, need not be the same for each modem, but may be varied to accommodate known disproportions in the data to be transmitted by A and B modems.

According to another aspect of the invention, signal loss and frequency offset are measured prior to data determination. A tracking system determines variations from the measured values and compensates for these deviations.

According to a further aspect of the invention, a system for determining a precise value of T_0 is included. This system utilizes two timing signals, at f_1 and f_2, incorporated in a waveform transmitted from modem A at time T_A. The relative phase difference between the first and second timing signals at time T_A is zero.

The waveform is received at modem B and a rough estimate, T_{EST}, of the time of reception is obtained by detecting energy at f_1. The relative phase difference between the timing signals at time T_{EST} is utilized to obtain a precise timing reference, T_0.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a graph of the ensemble of carrier frequencies utilized in the present invention.

Fig. 2 is a graph of the constellation illustrating the QAM of each carrier.

Fig. 3 is a block diagram of an embodiment of the invention.

Fig. 4 is a flow chart illustrating the synchronization process of the present invention.

Fig. 5 is a series of graphs depicting the constellations for 0, 2, 4, 5, 6 bit data elements and exemplary signal to noise ratios and power levels for each constellation.

Fig. 6 is a graph illustrating the waterfilling algorithm.

Fig. 7 is a histogram illustrating the application of the waterfilling algorithm utilized in the present invention.

Fig. 8 is a graph depicting the effects of phase dependent frequency delay on frequency components in the ensemble.

Fig. 9 is a graph depicting the wave forms utilized in the present invention to prevent intersymbol interference.

Fig. 10 is a graph depicting the method of receiving the transmitted ensemble.

Fig. 11 is a schematic diagram depicting the modulation template.

Fig. 12 is a schematic diagram depicting the quadrants of one square in the modulation template.

Fig. 13 is a schematic diagram of a hardware embodiment of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The present invention is a modem that adaptively allocates power between various carrier frequencies in a frequency ensemble to compensate for frequency dependent line noise, eliminates the need for equalization circuitry to compensate for a frequency dependent phase delay, and provides a duplex mechanism that accounts for varying channel load conditions to allocate the channel between the send and receive modems. Additional features of the invention are described below.

A brief description of the frequency ensemble and modulation scheme utilized in the present invention is first presented with respect to Figs. 1 and 2 to facilitate the understanding of the invention. A specific embodiment of the invention is then described with reference to Fig. 3. Finally, the operation of various features of the invention are described with reference to Figs. 4 through 13.

Modulation and Ensemble Configuration

Referring now to Fig. 1, a diagrammatic representation is shown of the transmit ensemble 10 of the present invention. The ensemble includes 512 carrier frequencies 12 equally spaced across the available 4 kHz VF band. The present invention utilizes quadrature amplitude modulation (QAM) wherein phase independent sine and cosine signals at each carrier frequency are transmitted. The digital information transmitted at a given carrier frequency is encoded by amplitude modulating the independent sine and cosine signals at that frequency.

The QAM system transmits data at an overall bit rate, R_B. However, the transmission rate on each carrier, denoted the symbol or baud rate, R_S, is only a fraction of R_B. For example, if data were allocated equally between two carriers then $R_S = R_B/2$.

In the preferred embodiment 0, 2, 4, 5 or 6 bit data elements are encoded on each carrier and the modulation of each carrier is changed every 136 msec. A theoretical maximum, R_B, assuming a 6 bit R_S for each carrier, of 22,580 bit/sec (bps) results. A typical relizable R_S, assuming 4 bit R_S over 75% of the carriers, is equal to about 11,300 bps. This extremely high R_S is achieved with a bit error rate of less than 1 error/100,000 bits transmitted.

In Fig. 1, a plurality of vertical lines 14 separates each ensemble into time increments known hereafter as "epochs." The epoch is of duration T_E where the magnitude of T_E is determined as set forth below.

The QAM system for encoding digital data onto the various carrier frequencies will now be described with reference to Fig. 2. In Fig. 2 a four bit "constellation" 20 for the nth carrier is depicted. A four bit number may assume sixteen discrete values. Each point in the constellation represents a vector (x_n, y_n) with x_n being the amplitude of the sine signal and y_n being the amplitude of the cosine signal in the above-described QAM system. The subscript n indicates the carrier being modulated. Accordingly, the four bit constellation requires four discrete y_n and four discrete x_n values. As described more fully below, increased power is required to increase the number of bits transmitted at a given carrier frequency due to the equivalent noise component at that frequency. The receive modem, in the case of four bit transmission, must be able to discriminate between four possible values of the x_n and y_n amplitude coefficients. This ability to discriminate is dependent on the signal to noise ratio for a given carrier frequency.

In a preferred embodiment, packet technology is utilized to reduce the error rate. A packet includes the modulated epoch of carriers and error detection data. Each packet in error is retransmitted until correct. Alternatively, in systems where retransmission of data is undesirable, epochs with forward error correcting codes may be utilized.

Block Diagram

Fig. 3 is a block diagram of an embodiment of the present invention. The description that follows is of an originate modem 26 coupled to an originate end of a communication link formed over a public switched telephone line. It is understood that a communication system also includes an answer modem coupled to the answer end of the communication link. In the following discussion, parts in the answer modem corresponding to identical or similar parts in the originate modem will be designated by the reference number of the originate modem primed.

Referring now to Fig. 3, an incoming data stream is received by a send system 28 of the modem 26 at data input 30. The data is stored as a sequence of data bits in a buffer memory 32. The output of buffer memory 32 is coupled to the input of a modulation parameter generator 34. The output of the modulation parameter generator 34 is coupled to a vector table buffer memory 36 with the vector table buffer memory 36 also coupled to the input of a modulator 40. The output of the modulator 40 is coupled to a time sequence buffer 42 with the time sequence buffer 42 also coupled to the input of a digital-to-analog converter 43 included in an analog I/O interface 44. The interface 44 couples the output of the modem to the public switched telephone lines 48.

A receive system 50 includes an analog-to-digital converter (ADC) 52 coupled to the public switched telephone line 48 and included in the interface 44. The output from the ADC 52 is coupled to a receive time series buffer 54 which is also coupled to the input of a demodulator 56. The output of the demodulator 56 is coupled to a receive vector table buffer 58 which is also coupled to the input of a digital data generator 60. The digital data generator 60 has an output coupled to a receive data bit buffer 62 which is also coupled to an output terminal 64.

A control and scheduling unit 66 is coupled with the modulation parameter generator 34, the vector table buffer 36, the demodulator 56, and the receive vector table buffer 58.

An overview of the functioning of the embodiment depicted in Fig. 3 will now be presented. Prior to the transmission of data, the originate modem 26, in cooperation with the answer modem 26', measures the equivalent noise level at each carrier frequency, determines the number of bits per epoch to be transmitted on each carrier frequency, and allocates power to each carrier frequency as described more fully below.

The incoming data is received at input port 30 and formatted into a bit sequence stored in the input buffer 32.

The modulator 34 encodes a given number of bits into an (x_n, y_n) vector for each carrier frequency utilizing the QAM system described above. For example, if it were determined that four bits were to be transmitted at frequency f_n then four bits from the bit stream would be converted to one of the sixteen points in the four bit constellation of Fig. 2. Each of these constellation points corresponds to one of sixteen possible combinations of four bits. The amplitudes of the sine and cosine signals for frequency n then corresponds to the point in the constellation encoding the four bits of the bit sequence. The (x_n, y_n) vectors are then stored in the vector buffer table 36. The modulator receives the table of (x_n, y_n) vectors for the carriers in the ensemble and generates a digitally encoded time series representing a wave form comprising the ensemble of QAM carrier frequencies.

In a preferred embodiment the modulator 40 includes a fast Fourier transform (FFT) and performs an inverse FFT operation utilizing the (x,y) vectors as the FFT coefficients. The vector table includes 1,024 independent points representing the 1,024 FFT points of the 512 frequency constellation. The inverse FFT operation generates 1,024 points in a time series representing the QAM ensemble. The 1,024 elements of this digitally encoded time series are stored in the digital time series buffer 42. The digital time sequence is converted to an analog wave form by the analog to digital converter 43 and the interface 46 conditions the signal for transmission over the public switched telephone lines 48.

Turning now to the receive system 50, the received analog waveform from the public switched telephone lines 48 is conditioned by the interface 46 and directed to the analog to digital converter 52. The analog to digital converter 52 converts the analog waveform to a digital 1,024 entry time series table which is stored in the receive time series buffer 54. The demodulator 56 converts the 1,024 entry time series table into a 512 entry (x_n, y_n) vector table stored in the receive vector table buffer 58. This conversion is accomplished by performing an FFT on the time series. Note that information regarding the number of bits encoded onto each frequency carrier has been previously stored in the demodulator and digital data generator 60 so that the (x,y) table stored in the receive vector table buffer 58 may be transformed to an output data bit sequence by the digital data generator 60. For example, if the (x_n, y_n) vector represents a four bit sequence then this vector would be converted to a four bit sequence and stored in the receive data bit buffer 62 by the digital data generator 60. The receive data bit sequence is then directed to the output 64 as an output data stream.

A full description of the FFT techniques utilized is described in a book by Rabiner et al., entitled Theory and Applications of Digital Signal Processing, Prentice-Hall, Inc., N.J., 1975. However, the FFT modulation technique described above is not an integral part of the present invention. Alternatively, modulation could be accomplished by direct multiplication of the carrier tones as described in the above-referenced Baran patent, which is hereby incorporated by reference, at col. 10, lines 13-70, and col. 11, lines 1-30. Additionally, the demodulation system described in Baran at col. 12, lines 35-70, col. 13, lines 1-70, and col. 14, lines 1-13 could be substituted.

The control and scheduling unit 66 maintains overall supervision of the sequence of operations and controls input and output functions.

Determination of Equivalent Noise

As described above, the information content of the data element encoded on each frequency carrier and the power allocated to that frequency carrier depends on the magnitude of the channel noise component at that carrier frequency. The equivalent transmitted noise component at frequency f_n, $N(f_n)$, is the measured (received) noise power at frequency f_n multiplied by the measured signal loss at frequency f_n. The equivalent noise varies from line to line and also varies on a given line at different times. Accordingly, in the present system, $N(f)$ is measured immediately prior to data transmission.

The steps of a synchronization technique utilized in the present system to measure $N(f)$ and establish a transmission link between answer and originate modems 26 and 26' are illustrated in Fig. 4. Referring now to Fig. 4, in step 1 the originate modem dials the number of the answer modem and the answer modem goes off hook. In step 2 the answer modem transmits an epoch of two frequencies at the following power levels:

(a) 1437.5 Hz. at -3 dBR; and

(b) 1687.5 Hz at -3 dBR.

The power is measured relative to a reference, R, where, in a preferred embodiment, 0dBR = -9dBm, m being a millivolt. These tones are used to determine timing and frequency offset as detailed subsequently.

The answer modem then transmits an answer comb containing all 512 frequencies at -27dBR. The originate modem receives the answer comb and performs an FFT on the comb. Since the power levels of the 512 frequencies were set at specified values, the control and scheduling unit 66 answer modem 26 compares the (x_n,y_n) values for each frequency of the received code and compares those values to a table of (x_n,y_n) values representing the power levels of the transmitted answer code. This comparison yields the signal loss at each frequency due to the transmission over the VF telephone lines.

During step 3 both the originate and answer modems 26 and 26' accumulate noise data present on the line in the absence of any transmission by either modem. Both modems then perform an FFT on the accumulated noise signals to determine the measured (received) noise spectrum component values at each carrier frequency. Several epochs of noise may be averaged to refine the measurement.

In step 4 the originate modem transmits an epoch of two frequencies followed by an originate comb of 512 frequencies with the same power levels described above for step 2. The answer modem receives the epoch and the originate comb and calculates the timing, frequency offset and signal loss values at each carrier frequency as described above for the originate modem in step 2. At this point the originate modem 26 has accumulated noise and signal loss data for transmission in the answer originate direction while the answer modem has accumulated the same data relating to transmission in the originate answer direction. Each modem requires data relating to transmission loss and receive noise in both the originate-answer and answer-originate directions. Therefore, this data is exchanged between the two modems according to the remaining steps of the synchronization process.

In step 5 the originate modem generates and transmits a first phase encoded signal indicating which carrier frequencies will support two bit transmission at standard power levels in the answer-originate direction. Each component that will support two bits in the answer-originate direction at a standard power level is generated as a -28 dBR signal with 180° relative phase. Each component that will not support two bit transmission in the answer-originate direction at the standard power level is coded as a -28 dBR,0° relative phase signal. The answer modem receives this signal and determines which frequency carriers will support two bit transmission in the answer-originate direction.

In step 6 the answer modem generates and transmits a second phase encoded signal indicating which carrier frequencies will support two bit transmission in both the originate-answer and answer-originate directions. The generation of this signal is possible because the answer modem has accumulated noise and signal loss data in the originate-answer direction and has received the same data for the answer-originate direction in the signal generated by the originate modem in step 5. In the signal generated by the originate modem, each frequency component that will support two bits in both directions is coded with 180° relative phase and all other components are coded with 0° relative phase.

A transmission link now exists between the two modems. In general, 300 to 400 frequency components will support two bit transmission at a standard power level, thereby establishing about a 600 bit/epoch rate between the two modems. In step 7 the originate modem sends data on the number of bits (0 to 15) and the power levels (0 to 63dB) that can be supported on each frequency in the answer-originate direction in ensemble packets formed over this existing data link. Accordingly, both the originate and answer modem now have the data relating to transmission in the answer-originate direction. The steps for calculating the number of bits and power levels that can be supported on each frequency component will be described below.

In step 8 the answer modem sends data on the number of bits and power levels that can be supported on each frequency in the originate-answer direction utilizing the existing data link. Thus, both modems are apprised of the number of bits and power levels to be supported on each frequency component in both the answer-originate and originate-answer directions.

The above description of the determination of the equivalent noise level component at each carrier frequency sets forth the required steps in a given sequence. However, the sequence of steps is not critical and many of the steps may be done simultaneously or in different order, for example, the performance of the FFT on the originate code and the accumulation of noise data may be done simultaneously. A precise timing reference is also calculated during the synchronization process. The calculation of this timing reference will be described more fully below after the description of the method for calculating the number of bits and power levels allocated to each frequency component.

It is a common VF telephone line impairment that a frequency offset, of up to 7 Hz, exists between transmitted and received signals. This offset must be corrected for the FFT to function reliably. In a preferred embodiment, this correction is achieved by performing a single sideband modulation of the quadrature tones at the offset frequency by the true and Hilbert images of received signal. Synchronization and tracking algorithms generate estimates of the frequency offset necessary.

Power and Code Complexity Allocation

The information encoded on each carrier frequency signal is decoded at the receiver channel by the demodulator 56. Channel noise distorts the transmitted signal and degrades the accuracy of the demodulation process. The transmission of a data element having a specified complexity, e.g., B_0 bits at a specified frequency, f_0, over a VF telephone line characterized by an equivalent noise level component, N_0, will now be analyzed. Generally, external system requirements determine a maximum bit error rate (BER) that can be tolerated. For the transmission of b_0 bits at noise level N_0 and frequency f_0, the signal to noise ratio must exceed E_b/N_0 where E_b is the signal power per bit to maintain the BER below a given BER, $(BER)_0$.

Fig. 5 depicts the QAM constellations for signals of various complexities B. An exemplary signal to noise ratio, E_b/N_0, for each constellation and the power required to transmit the number of bits in the constellation without exceeding $(BER)_0$ is depicted alongside each constellation graph.

A modem operates under the constraint that the total available power placed on the public switched telephone lines may not exceed a value, P_0, set by the telephone companies and government agencies. Thus, signal power may not be increased indefinitely to compensate for line noise. Accordingly, as noise increases, the complexity of the signals transmitted must be decreased to maintain the required BER.

Most existing modems arbitrarily gear shift the signal complexity down as line noise power increases. For example, one prior art modem reduces the transmitted data rate from a maximum of 9,600 bps to steps of 7,200 bps, 4,800 bps, 2,400 bps, 1,200 bps, and so on until the bit error rate is reduced below a specified maximum. Accordingly, the signal rate is decreased in large steps to compensate for noise. In the Baran patent, the method for reducing the transmission rate takes into account the frequency dependent nature of the noise spectrum. There, each channel carries a preset number of bits at a specified power level. The noise component at each frequency is measured and a decision is made whether to transmit at each carrier frequency. Thus, in Baran, the data rate reduction scheme compensates for the actual distribution of the noise over the available bandwidth.

In the present invention, the complexity of the signal on each frequency carrier and the amount of the available power allocated to each frequency carrier is varied in response to the frequency dependence of the line noise spectrum.

The present system for assigning various code complexities and power levels to the frequency component signals in the ensemble is based on the waterfilling algorithm. The waterfilling algorithm is an information theoretic way of assigning power to a channel to maximize the flow of information across the channel. The channel is of the type characterized by an uneven noise distribution and the transmitter is subject to a power constraint. Fig. 6 provides a visualization of the waterfilling algorithm. Referring now to Fig. 6, power is measured along the vertical axis and frequency is measured along the horizontal axis. The equivalent noise spectrum is represented by the solid line 70 and the available power is represented by the area of the cross hatched region 72. The name waterfilling comes from the analogy of the equivalent noise function to a series of valleys in a mountain filled with a volume of water representing the assigned power. The water fills the valleys and assumes a level surface. A theoretical description of the waterfilling algorithm is given in the book by Gallagher, entitled <u>Information Theory And Reliable Communication</u>; J. Wiley and Sons, New York, 1968, p. 387.

It must be emphasized that the waterfilling theorem relates to maximizing the theoretical capacity of a channel where the capacity is defined as the maximum of all data rates achievable using different codes, all of which are error correcting, and where the best tend to be of infinite length.

The method utilizing the present invention does not maximize the capacity of the channel. Instead, the method maximizes the amount of information transmitted utilizing the QAM ensemble described above with respect to Fig. 1 and subject to an available power restriction.

An implementation of the waterfilling concept is to allocate an increment of available power to the carrier having the lowest equivalent noise floor until the allocatd power level reaches the equivalent noise level of the second lowest carrier. This allocation requires a scan through the 512 frequencies.

Incremental power is then allocated between the lowest two carriers until the equivalent noise level of the third lowest channel is reached. This allocation level requires many scans through the frequency table and is computationally complex.

The power allocation method used in a preferred embodiment of the present invention is as follows:

(1) Calculate the system noise at the transmitter by measuring the equivalent noise at the receiver and multiplying by transmission loss. This process for measuring these quantities was described above with respect to synchronization and Fig. 4. The system noise components are calculated for each carrier frequency.

(2) For each carrier frequency, calculate the power levels required to transmit data elements of varying complexity (in the present case, 0, 2, 4, 5, 6, and 8 bits). This is accomplished by multiplying the equivalent noise by the signal to noise ratios necessary for transmission of the various data elements with a required BER, for example one error per 100,000 bits. The overall BER is the sum of the signal error rates of each modulated carrier. These signal to noise ratios are available from standard references, and are well-known in the art.

(3) From the calculated required transmission power levels, the marginal required power levels to increase data element complexity are determined. These marginal required power levels are the difference in transmission power divided by the quantitative difference in complexity of the data elements closest in complexity.

(4) For each channel generate a two column table of marginal required power levels and quantitative differences where the units are typically expressed as Watts and bits, respectively.

(5) Construct a histogram by organizing the table of step 4 according to increasing marginal power.

(6) Assign the available transmitter power sequentially over the increasing marginal powers until available power is exhausted.

The power allocation method may be better understood through a simple example. The numbers presented in the example are not intended to represent parameters encountered in an operating system.

Table 1 sets out the power requirement, P, to transmit a data element of a selected number of bits, N_1, for two carriers A and B at frequencies f_A and f_B.

TABLE 1

Carrier A

N_1	N_2-N_1	P	MP(N_1 to N_2)
0	-	0	-
2	2	4	MP(0to2)=2/bit
4	2	12	MP(2to4)=4/bit
5	1	19	MP(4to5)=7/bit
6	1	29	MP(5to6)=10/bit

Carrier B

N_1	N_2-N_1	P	MP(N_1 to N_2)
0	-	0	-
2	2	6	MP(0to2)=3/bit
4	2	18	MP(2to4)=6/bit
5	1	29	MP(4to5)=11/bit
6	1	44	MP(5to6)=15/bit

The marginal power to increase the complexity from a first number of bits, N_1, to a second number of bits, N_2, is defined by the relationship:

$$MP(N_1 \text{ to } N_2) = \frac{P_2 - P_1}{N_2 - N_1}$$

where P_2 and P_1 are the powers required to transmit data elements of complexity N_2 and N_1. N_2-N_1 is quantitative difference in the complexity of the data elements. It is understood the BER is constrained to remain below a preset limit.

The marginal powers for f_A are less than for f_B because the equivalent noise at f_B, $N(f_B)$, is greater than the equivalent noise at f_A, $N(f_A)$.

The implementation of the allocation scheme for carriers A and B will now be described. Assume that a total number of bits, N_T, are encoded on the ensemble but that no bits have been assigned to carriers A or B. For example, $N(f_A)$ and $N(f_A)$ might be greater than the powers of those carriers already carrying the data.

In this example, the system is to allocate ten remaining available power units between carriers A and B to increase the overall data element complexity by the maximum amount.

To increase N_T by two bits requires that four units of power be allocated if channel A is utilized and that six units of power be allocated in channel B is utilized. This follows because for both channels $N_1 = 0$ and $N_2 = 2$ and MP(0 to 2) = 2/bit for channel A and MP(0 to 2) = 3/bit for channel B. Therefore, the system allocates four units of power to carrier A, encodes a two bit data element on carrier A, increases the overall signal complexity from N_T to $N_T + 2$, and has six remaining available power units.

The next increase of two bits requires six power units because MP(2 to 4) = 4/bit for carrier A and MP(0 to 2) = 3/bit for channel B. Therefore, the system allocates six units of power to carrier B, encodes a two bit data element on carrier B, increases the overall signal complexity from $N_T + 2$ to $N_T + 4$ bits, and has no remaining available power units.

As is now clear, the system "shops" among the various carrier frequencies for the lowest power cost to increase the complexity of the overall ensemble data element.

The allocation system is extended to the full 512 carrier ensemble by first generating the tables of

Table 1 for each carrier during a first pass through the frequencies.

A histogram organizing the calculated marginal required power levels for all the carriers according to increasing power is then constructed. Fig. 7 is a depiction of an exemplary histogram constructed according to the present method.

In Fig. 7 the entire table of marginal powers is not displayed. Instead, the histogram is constructed having a range of 64dB with counts spaced in 0.5dB steps. The quantitative differences between the steps are utilized as counts. Although this approach results in a slight round-off error, a significant reduction in task length is achieved. The method used to construct the histogram is not critical to practicing the invention.

Each count of the histogram has an integer entry representing the number of carriers having a marginal power value equal to the power value at the count. The histogram is scanned from the lowest power level. The integer entry at each count is multiplied by the number of counts and subtracted from the available power. The scan continues until available power is exhausted.

When the scan is completed it has been determined that all marginal power values below a given level, MP(max), are acceptable for power and data allocation. Additionally, if available power is exhausted partially through marginal power level, MP(max), then k additional carriers will be allocated power equal to MP(max + 1).

The system then scans through the ensemble again to allocate power and data to the various carriers. The amount of power allocated to each carrier is the sum of marginal power values for that carrier less than or equal to MP(max). Additionally, an amount of power equal to MP(max + 1) will be allocated if the k MP(max + 1) values have not been previously allocated.

Timing and Phase Delay Compensation

The reconstruction of (x,y) vector table by the receive system requires 1024 time samples of the received waveform. The bandwidth is about 4kHz so that Nyquist sampling rate about 8000/sec and the time sample offset between samples is 125 microseconds. The total sampling time is thus 128 msec. Similarly, the transmit FFT generates a time series having 1024 entries and the symbol time is 128 msec.

The sampling process requires a timing reference to initiate the sampling. This timing reference is established during synchronization by the following method:

During the synchronization steps defined with reference to Fig. 4, the originate modem detects energy at the 1437.5 Hz frequency component (the first timing signal) in the answer comb at time T_{EST}. This time is a rough measure of the precise time that the first timing frequency component arrives at the receiver and is generally accurate to about 2 msec.

This rough measure is refined by the following steps. The first timing signal and a second timing signal (at 1687.5 Hz) are transmitted with zero relative phase at the epoch mark.

The originate modem compares the phases of the first and second timing signals at time T_{EST}. The 250 Hz frequency difference between the first and second timing signals results in an 11° phase shift between the two signals for each 125 microsecond time sample offset. The first and second timing signals have low relative phase distortion (less than 250 microseconds) due to their location near the center of the band. Accordingly, by comparing the phases of the two timing samples and correcting T_{EST} by the number of time sampling offsets indicated by the phase difference, a precise timing reference, T_0, can be determined.

A further difficulty relating to timing the sampling process relates to frequency dependent phase delay induced by the VF line. This phase delay typically is on the order of 2 msec, or more, for VF telephone lines. Further, this phase delay is significantly worse near the edges of the 4kHz usable band.

Fig. 8 depicts distribution of the frequency carriers of the ensemble after undergoing frequency dependent phase delay. Referring to Fig. 8, three signals 90, 92, and 94 at frequencies f_0, f_{256}, and f_{512} are depicted. Two symbols, x_i and y_i, of length T_S are transmitted at each frequency. Note that the duration of each symbol is not changed. However, the leading edge of the signals near the edge of the band 92 and 94 are delayed relative to those signals near the center of the band 94.

Additionally, for two sequentially transmitted epochs x_i and y_i the trailing section of the first symbol x_i on signals 92 and 96, near the outer edge of the band will overlap the leading edge of the second symbol y_i on the signal 94 near the center of the band. This overlap results in intersymbol interference.

If the sampling interval is framed to sample a given time interval, T_S, then complete samples of every carrier in the ensemble will not be obtained and signals from other epochs will also be sampled.

Existing systems utilize phase correction (equalization) networks to correct for phase distortion and to prevent intersymbol interference.

The present invention utilizes a unique guard-time format to eliminate the need for an equalization network. This format is illustrated in Fig. 9.

Referring now to Fig. 9, first, second, and third transmitted symbols, represented by time series x_i, y_i, and z_i, respectively, are depicted. The waveforms depicted in Fig. 3 are modulated on one of the carriers at frequency f. In this example a symbol time, T_S, of 128 msec. and a maximum phase delay, T_{PH}, of 8 msec are assumed. A guard-time waveform is formed by repeating the first 8 msec. of the symbol. The guard-time waveform defines an epoch of 136 msec. For example, in the first waveform 110, (X_i), the time series of the symbol, $X_0 - X_{1023}$, is first transmitted, then the first 8 msec. of the symbol, $X_0 - X_{63}$, are repeated.

The sampling of the epoch is aligned with the <u>last</u> 128 msec. of the guard-time waveform (relative to the beginning of the guard-time epoch defined by those frequency components which arrive first).

This detection process is illustrated in Fig. 10. In Fig. 10 first and second guard-time waveforms 110 and 112 at f_1, near the center of the band, and f_2, near the edge of the band, are depicted. The frequency component at f_1 is the component of the ensemble that arrives first at the receiver and the component at f_2 arrives last. In Fig. 10 the second waveform 112, at f_2, arrives at the receiver at $T_0 + T_{PH}$, which is 8 msec. after the time, T_0, that the first waveform 110, at f_1, arrives at the receiver. The sampling period of 128 msec. is initiated at the time $T_0 + T_{PH}$. Thus, the entire symbol on f_2, $X_0 - X_{1023}$, is sampled. The entire symbol at f_1 is also sampled because the initial 8 msec. of that symbol has been retransmitted.

Also, intersymbol interference has been eliminated. The arrival of the second symbol, (y_i), at f_1 has been delayed 8 msec. by the retransmission of the first 8 msec. of (x_i). Thus, the leading edge of the second symbol at f_1, does not overlap the trailing edge of the first symbol at f_2.

The 8 msec. guardtime reduces the usable time-bandwidth product of the system by only about 6%. This small decrease is due to the very long duration of each symbol relative to the necessary guardtime.

Tracking

In practice, for a given carrier, the magnitudes of the (x,y) vectors extracted during the demodulation process do not fall exactly at the constellation points but are distributed over a range about each point due to noise and other factors. Accordingly, the signal is decoded utilizing a modulation template as depicted in Fig. 11.

Referring now to Fig. 11, the template is formed by a grid of squares 113 with the constellation points 114 at the centers of the squares 113.

In Fig. 11, the vector $W = (x_n, y_n)$ represents the demodulated amplitudes of the sine and cosine signals at f_n. W is in the square 113 having the constellation point (3,3) centered therein. Accordingly, W is decoded as (3,3).

The present invention includes a system for tracking to determine changes in transmission loss, frequency offset, and timing from the values determined during synchronization.

This tracking system utilizes the position of the received vectors in the squares of the demodulation template of Fig. 11. In Fig. 12, a single square is divided into four quadrants upper left, lower right, upper right, lower right, 115, 116, 117, and 118 characterized as too fast, too slow, too big, and too little, respectively. If counts in all four quadrants over time by frequency or over frequency at one time are equal or nearly equal then the system is in alignment. That is, if noise is the only impairment, then the direction of error for the decoded vector, W, should be random.

However, if transmission loss changes by even 0.1dB the number of too small counts will vary significantly from the number of too large counts. Similarly, a large difference between the number of too fast and too slow counts indicates a phase rotation caused by a change in the offset frequency. Thus, the differences between the too fast, too slow, and too big, too small counts is an error characteristic that tracks variations in signal loss and offset frequency.

The present invention utilizes this error characteristic to adjust the signal loss and frequency offset determined during synchronization. For each frequency an adjustment of ± .1dB or ± 1.0° is made depending on the error characteristic. Other divisions of the decoding region into distinct or overlapping subregions characterized as too fast, too slow, too big, and too little are preferred in some embodiments.

Additionally, the phase of the timing signals is tracked to allow corrections of T_0.

Allocation of Channel Control

The present invention further includes a unique system for allocating control of an established communication link between the originate and answer modems (hereinafter designated A and B, respectively). Each waveform comprising the encoded ensemble of frequencies forms a packet of information.

Control of the transmission link is first allocated to modem A. Modem A then determines the volume of data in its input buffer and transmits between I (a minimum) and N (a previously determined maximum) packets of data as appropriate. The predetermined number N serves as a limit and the end number of transmitted packets may be significantly less than required to empty the input buffer. On the other hand, if modem A has little or no data in its input buffer it will still transmit I packets of information to maintain communication with modem B. For example, the I packets may comprise the originate or answer comb of frequencies defined above with respect to Fig. 4 and the synchronization process.

Control of the communication link is then allocated to modem B which repeats the actions of modem A. Of course, if modem B transmits the minimum number, I, of packets it is confirming to modem A the vitality of modem B.

There is no need for the limits N on the two modems to be the same, or to restrict them from being adaptable under modem control to obtain rapid character echo or other user oriented goals.

Hardware Implementation

Fig. 13 is a block diagram of a hardware embodiment of the invention. Referring now to Fig. 13, an electronic digital processor 120, an analog I/O interface 44, and a digital I/O interface 122 are coupled to a common data bus 124. The analog I/O interface 44 interfaces the public switched telephone line 48 with the common data bus 124 and the digital interface 122 interfaces digital terminal equipment 126 with the common data bus 124.

The following components are utilized in a preferred embodiment of the invention. The analog I/O interface 44 is a high performance 12 bit coder-decoder (codec) and telephone line interface. The interface has access to RAM 132 and is controlled by supervisory microprocessor 128. The codec is a single chip combination of an analog to digital converter, a digital to analog converter, and several band pass filters.

The digital I/O interface 122 is a standard RS-232 serial interface to a standard twenty-five pin RS-232 type connector or a parallel interface to a personal computer bus.

The electronic digital processor 120, includes a supervisory processor 128, a general purpose mathematical processor 130, a 32K by 16 bit shared RAM subsystem 132, and a read only memory (ROM) unit 133, coupled to an address bus 135.

The supervisory microprocessor 128 is a 68000 data processor subsystem including a 10MHz 68000 processor and the 68000 program memory. The 32K by 16 bit program memory consists of several low power, high density, ROM chips included in the ROM unit 133.

The mathematical processor 130 is a 320 digital signal microprocessor system (DSP) including a 20MHz 320 processor, the 320 program memory, and an interface to the shared RAM system. Two high speed ROM chips, included in ROM unit 133, comprise the 8192 x 16 bit program memory.

The 320 system program memory includes programs for performing the modulation table look-up, FFT, demodulation, and other operations described above. The 68000 processor handles digital data streams at the input and output, performs tasking to and supervision of the 320 signal processor and associated analog I/O, and performs self and system test as appropriate.

The invention has been explained with respect to specific embodiments. Other embodiments will now be apparent to those of ordinary skill in the art.

In particular, the ensemble of carrier frequencies need not be limited as above-described. The number of carriers may be any power of 2, e.g. 1024, or some arbitrary number. Additionally, the frequencies need not be evenly spaced over the entire VF band. Further, the QAM scheme is not critical to practicing the invention. For example, AM could be utilized although the data rate, R_B, would be reduced.

Still further, the modulation template need not be comprised of squares. Arbitrarily shaped regions surrounding the constellation points may be defined. The tracking system was described where the squares in the modulation template were divided into four quadrants. However, a given parameter may be tracked by tracking the difference in the number of counts in arbitrary regions defined about a constellation point.

Still further, a hardware embodiment including a supervisory microprocessor and a general purpose mathematical processor has been described. However, different combinations of IC chips may be utilized. For example, a dedicated FFT chip could be utilized to perform modulation and demodulation operations.

Still further, the information units utilized in the above description were bits. However, the invention is not limited to binary system.

Accordingly, it is therefore intended that the invention can be limited except as indicated by the appended claims.

WHAT IS CLAIMED IS:

1. In a high speed modem, for transmitting data over a telephone line, of the type that encodes data elements on an ensemble of carrier frequencies, a method for allocating data and power to the carrier frequencies, said method comprising the steps of:

determining the equivalent noise component for every carrier frequency in the ensemble;

determining the marginal power requirement to increase the complexity of the data element on each carrier from n information units to n + 1 information units, n being an integer between 0 and N;

ordering the marginal powers of all the carriers in the ensemble in order of increasing power;

assigning available power to the ordered marginal powers in order of increasing power;

determining the value, MP(max) at which point the available power is exhausted; and

allocating power and data to each carrier frequency where the power allocated is equal to the sum of all the marginal powers less than or equal to MP(max) for that carrier and the number of data units allocated is equal to the number of marginal powers for that carrier less than or equal to MP(max).

2. The invention of claim 1 where said step of ordering comprises the steps of:

providing a table of arbitrary marginal power levels; and

rounding the value of each determined marginal power level to one of the values of the table of arbitrary marginal power levels to decrease computational complexity.

3. The invention of claim 2 wherein the step of determining equivalent noise comprises the steps of:

providing an A and a B modem interconnected by a telephone line;

establishing a communication link between said A and B modems;

accumulating line noise data during a no transmission time interval at said A and B modems;

transmitting at least a first ensemble of frequency carriers from said A modem to said B modem, where the amplitude of each carrier has a predetermined value;

receiving said first ensemble at said B modem;

measuring the amplitude of each carrier received at said B modem;

comparing the measured amplitudes at said B modem with said predetermined amplitudes to determine signal loss, in dB, at each carrier frequency;

determining the value of the component, in dB, at each carrier frequency of the accumulated noise; and

adding the signal loss at each carrier frequency to the noise component at each carrier frequency to determine equivalent noise.

4. A high speed modem of the type for transmitting a signal on a VF telephone line, comprising:

means for receiving an input digital data stream and for storing said input digital data;

means for generating a modulated ensemble of carriers to encode said input digital data, where each carrier has data elements of variable complexity encoded thereon;

means for measuring the signal loss and noise loss of the VF telephone line for each carrier; and means for varying the complexity of the data element encoded on each carrier and the amount of power allocated to each carrier to compensate for the measured signal loss and noise level.

5. A high speed modem of the type that encodes data elements on an ensemble of carriers of different frequency, said modem comprising:

a digital electronic processor;

a digital electronic memory;

bus means for coupling said processor and said memory;

means, associated with said digital electronic processor, for

determining the equivalent noise component for every carrier frequency in the ensemble;

determining the marginal power requirements to increase the complexity of the data element on each carrier from n information units to n + 1 information units, n being an integer between 0 and N;

ordering the marginal powers of all the carriers in the ensemble in order of increasing power;

assigning available power to the ordered marginal powers in order of increasing power;

determining the value, MP(max) at which point the available power is exhausted; and

assigning power and data to each carrier frequency where the power assigned is equal to the sum of all the marginal powers less than or equal to MP(max) for that carrier and the number of data units is equal to the number of marginal powers for that carrier less than or equal to MP(max).

6. In a high speed modem, for transmitting data in the form of a QAM ensemble of carrier frequencies on a VF telephone line, of the type that measures the magnitude of a system parameter prior to

transmission, a method for tracking deviations in the magnitude of the system parameter during the receipt of data, said method comprising the steps of:

generating QAM constellations for a plurality of carrier frequencies;

constructing a demodulation template for one of said plurality of carrier frequencies comprising a plurality of first regions with one of the points of said constellation positioned within each of said first regions;

forming a set of tracking regions where each first region has a first and second tracking region disposed therein;

demodulating said ensemble of carriers to obtain the demodulation points positioned in said set of first and second tracking regions;

counting the number of points disposed in said set of first tracking regions and the number of points disposed in said set of second tracking regions;

determining the difference in the number of counts disposed in said set of first tracking regions and disposed in said tracking regions to construct an error characteristic; and

utilizing said error characteristic to adjust the magnitude of said signal parameter during the receipt of data.

7. The invention of claim 6 wherein said step of constructing a demodulation template comprises the step of:

constraining said first regions to be in the shape of squares having said constellation points centered therein.

8. The invention of claim 7 wherein said step of forming said tracking regions comprises the step of:

dividing said squares into quadrants; and

selecting said tracking regions to be symmetrically disposed quadrants.

9. In a communication system of the type including two modems (A and B) coupled by a transmission link, each modem having an input buffer for storing data to be transmitted, a method for allocating control of the transmission link between modem A and B comprising the steps of:

allocating control of the transmission link to modem A;

determining the volume of data stored in the input buffer of modem A;

determining the number, K, of packets of data required to transmit the volume of data stored in the input buffer of modem A;

transmitting L packets of data from modem A to modem B where L is equal to I_A if K is less than I_A, where L is equal to K if K is greater than or equal to I_A, and where L is equal to N_A if K is greater than N_A so that the minimum number of packets transmitted is I_A and the maximum is N_A;

allocating control of the transmission link to modem B;

determining the volume of data in the input buffer of modem B;

determining the number, J, of packets of data required to transmit the volume of data stored in the input buffer of modem B;

transmitting M packets of data from modem B to modem A where M is equal to I_B if J is less than I_B, where M is equal to J if J is greater than or equal to I_B, and where L is equal to N_B if J is greater than N_B so that the minimum number of packets transmitted is I_B and the maximum is N_B;

where allocation of control between modem A and B is dependent on the volume of data stored in the input buffers of modems A and B.

10. In a high speed modem, for transmitting data over a telephone line, of the type that encodes data elements on an ensemble of carrier frequencies, a system for allocating data and power to the carrier frequencies, said system comprising:

means for determining the equivalent noise component for every carrier frequency in the ensemble;

means for determining the marginal power requirement to increase the complexity of the data element on each carrier from n information units to n + 1 information units, n being an integer between 0 and N;

means for ordering the marginal powers of all the carriers in the ensemble in order of increasing power;

means for assigning available power to the ordered marginal powers in order of increasing power;

means for determining the value, MP(max) at which point the available power is exhausted; and

means allocating power and data to each carrier frequency where the power allocated is equal to the sum of all the marginal powers less than or equal to MP(max) for that carrier and the number of data units allocated is equal to the number of marginal powers for that carrier less than or equal to MP(max).

11. The invention of claim 10 where said means for ordering comprises:

means for providing a table of arbitrary marginal power levels; and

means for rounding the value of each determined marginal power level to one of the values of the table of arbitrary marginal power levels to decrease computational complexity.

12. The invention of claim 11 wherein an A and B modem are connected by a telephone line and the means for determining equivalent noise comprises:

means for establishing a communication link between said A and B modems;

means for accumulating line noise data during a no transmission time interval at said A and B modems;

means for transmitting a first ensemble of frequency carriers from said A modem to said B modem, where the amplitude of each carrier has a predetermined value;

means for receiving said first ensemble at said B modem;

means for measuring the amplitude of each carrier received at said B modem;

means for comparing the measured amplitudes at said B modem with said predetermined amplitudes to determine signal loss at each carrier frequency;

means for determining the value of the component, in dB, at each carrier frequency of the accumulated noise; and

means for adding the signal loss at each carrier frequency to the noise component at each carrier frequency to determine equivalent noise.

13. In a high speed modem, for transmitting data in the form of a QAM ensemble of carrier frequencies on a VF telephone line, of the type that measures the magnitude of a system parameter prior to transmission, a system for tracking deviations in the magnitude of the system parameter during the receipt of data, said system comprising:

means for generating QAM constellations for a plurality of carrier frequencies;

means for constructing a demodulation template for one of said plurality of carrier frequencies comprising a plurality of first regions with one of the points of said constellation positioned within each of said first regions;

means for forming a set of tracking regions where each first region has a first and second tracking region disposed therein;

means for demodulating said ensemble of carriers to obtain the modulation points positioned in said set of first and second tracking regions;

means for counting the number of points disposed in said set of first tracking regions and the number of points disposed in said set of second tracking regions;

means for determining the difference in the number of counts disposed in said set of first tracking regions and disposed in said tracking regions to construct an error characteristic; and

means for utilizing said error characteristic to adjust the magnitude of said signal parameter during the receipt of data.

14. The invention of claim 13 wherein said means for constructing a demodulation template comprises:

means for constraining said first regions to be in the shape of squares having said constellation points centered therein.

15. The invention of claim 14 wherein said means for forming said tracking regions comprises:

means for dividing said squares into quadrants; and

means for selecting said tracking regions to be symmetrically disposed quadrants.

16. In a communication system of the type including two modems (A and B) coupled by a transmission link, each modem having an input buffer for storing data to be transmitted, a system for allocating control

of the transmission link between modem A and B comprising:

means for allocating control of the transmission link to modem A;

means for determining the number, K, of packets of data required to transmit the volume of data stored in the input buffer of modem A;

means for transmitting L packets of data from modem A to modem B where L is equal to I_A if K is less than I_A but less than N_A, where L is equal to K if K is greater than or equal to I_A, and where L is equal to N_A if K is greater than N_A so that the minimum number of packets transmitted is I_A and the maximum is N_A;

means for allocating control of the transmission link to modem B;

means for determining the volume of data in the input buffer of modem B;

means for determining the number, J, of packets of data required to transmit the volume of data stored in the input buffer of modem B;

means for transmitting M packets of data from modem B to modem A where M is equal to I_B if J is less than I_B, where M is equal to J if J is greater than or equal to I_B but less than N_B, and where L is equal to N_B if J is greater than N_B so that the minimum number of packets transmitted is I_B and the maximum is N_B;

where allocation of control between modem A and B is dependent on the volume of data stored in the input buffers of modems A and B.

17. In a high speed modem communication system including two modems (A and B) coupled by a transmission link, each modem having an input buffer for storing data to be transmitted, each modem for transmitting data over a telephone line and each modem of the type that encodes data elements on an ensemble of carrier frequencies, a method of operating said modems to effi-

ciently allocate power and data to the carrier frequencies, to compensate for frequency dependent phase delay, where the maximum estimated magnitude of the phase delay is T_{PH}, to prevent intersymbol interference, to allocate control of the transmission link between modem A and modem B and for initiating a sampling interval having a given time sample offset equal to the reciprocal of the sampling frequency, said method comprising:

determining the equivalent noise component for every carrier frequency in the ensemble;

determining the marginal power requirement to increase the complexity of the data element on each carrier from n information units to n + 1 information units, n being an integer between 0 and N;

ordering the marginal powers of all the carriers in the ensemble in order of increasing power;

assigning available power to the ordered marginal powers in order of increasing power;

determining the value, MP(max) at which point the available power is exhausted;

allocating power and data to each carrier frequency where the power allocated is equal to the sum of all the marginal powers less than or equal to MP(max) for that carrier and the number of data units allocated is equal to the number of marginal powers for that carrier less than or equal to MP(max);

transmitting a symbol encoded on one of said carrier frequencies where said symbol is a predetermined time duration, T_S;

retransmitting the first T_{PH} seconds of said symbol to form a transmitted waveform of duration $T_E + T_{PH}$;

allocating control of the transmission link to modem A;

determining the volume of data stored in the input buffer of modem A;

determining the number, K, of packets of data required to transmit the volume of data stored in the input buffer of modem A;

transmitting L packets of data from modem A to modem B where L is equal to I_A if K is less than I_A, where L is equal to K if K is greater than or equal to I_A, and where L is equal to N_A if K is greater than N_A so that the minimum number of packets transmitted is I_A and the maximum is N_A;

allocating control of the transmission link to modem B;

determining the volume of data in the input buffer of modem B;

determining the number,J, of packets of data required to transmit the volume of data stored in the input buffer of modem B;

transmitting M packets of data from modem B to modem A where M is equal to I_B if J is less than I_B, where M is equal to J if J is greater than or equal to I_B, and where L is equal to N_B if J is greater than N_B so that the minimum number of packets transmitted is I_B and the maximum is N_B;

where allocation of control between modem A and B is dependent on the volume of data stored in the input buffers of modems A and B;

generating an analog waveform at modem A including first and second frequency components at f_1 and f_2;

transmitting said waveform from modem A to modem B at time T_A;

adjusting the phases of said first and and second frequency components so that their relative phase difference at time T_A is equal to about 0°;

detecting energy at frequency f_1 at modem B to determine the estimated time, T_{EST}, that said waveform arrives at modem B;

determining the relative phase difference at modem B between said first and second frequency components at time T_{EST};

calculating the number of sampling time offsets, N_I, required for the relative phase of said first and second carriers to change from 0 to said relative phase difference; and

changing the magnitude of T_{EST} by N_I sampling intervals to obtain a precise timing reference, T_0.

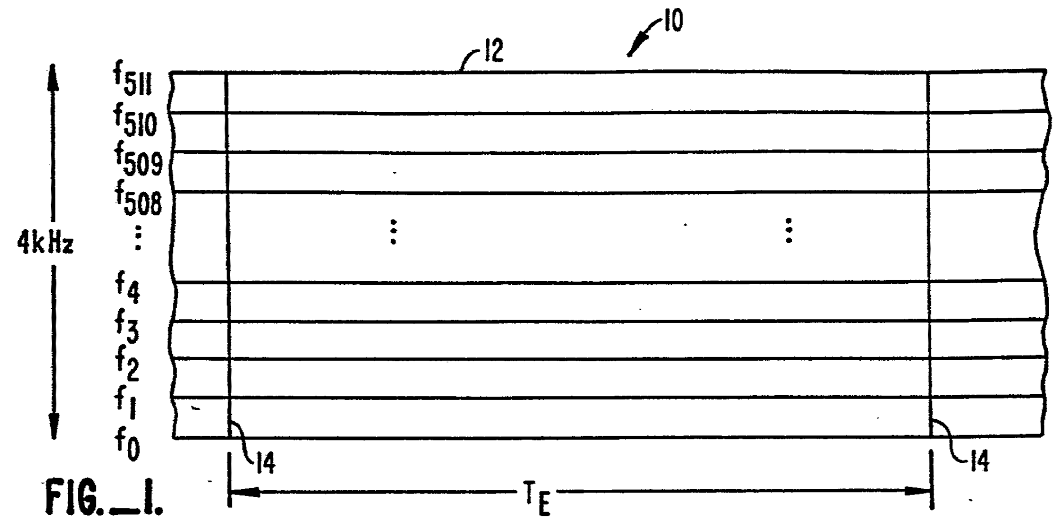


FIG._1.

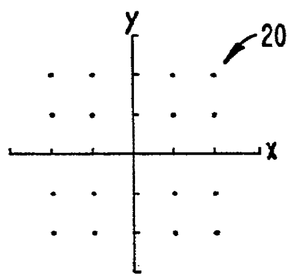


FIG._2.

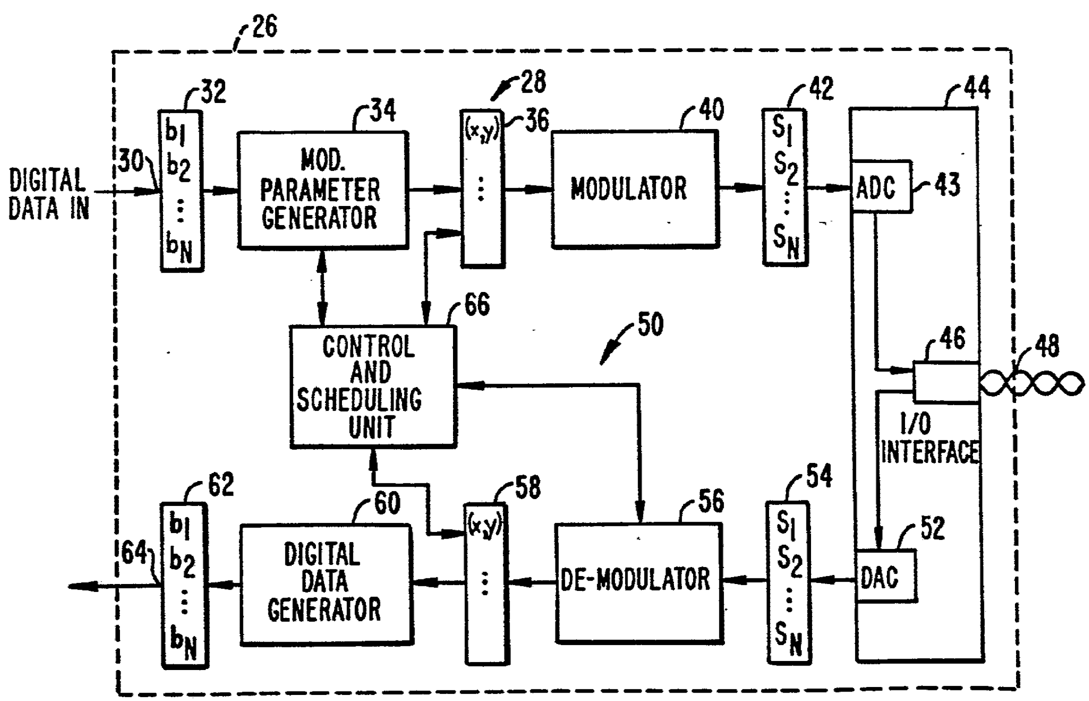


FIG._3.

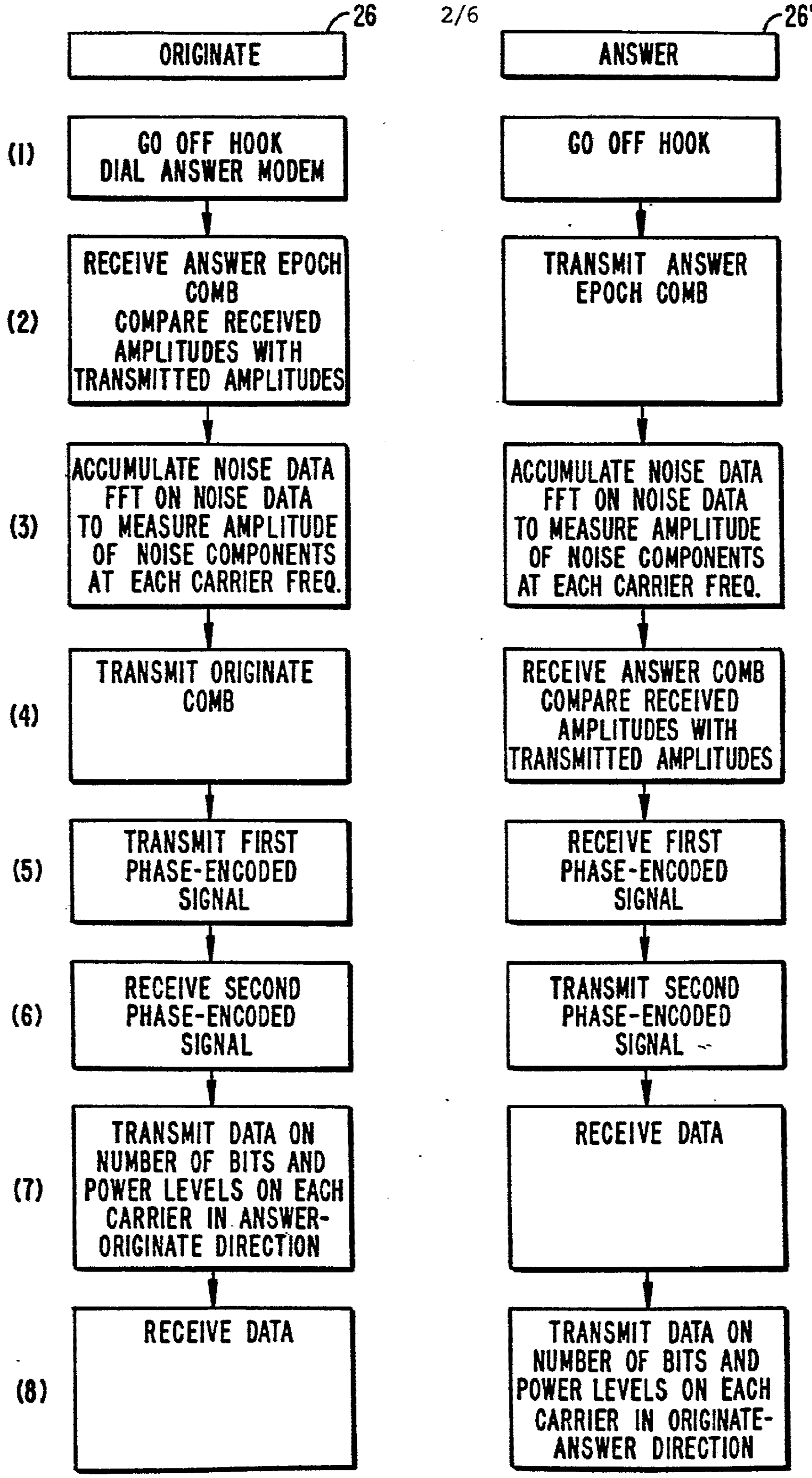
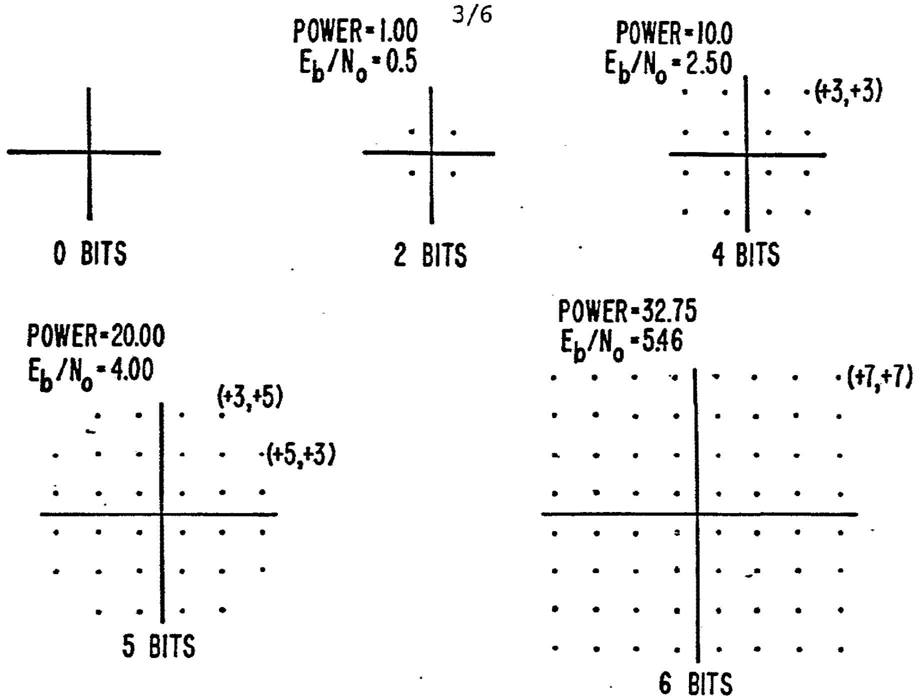
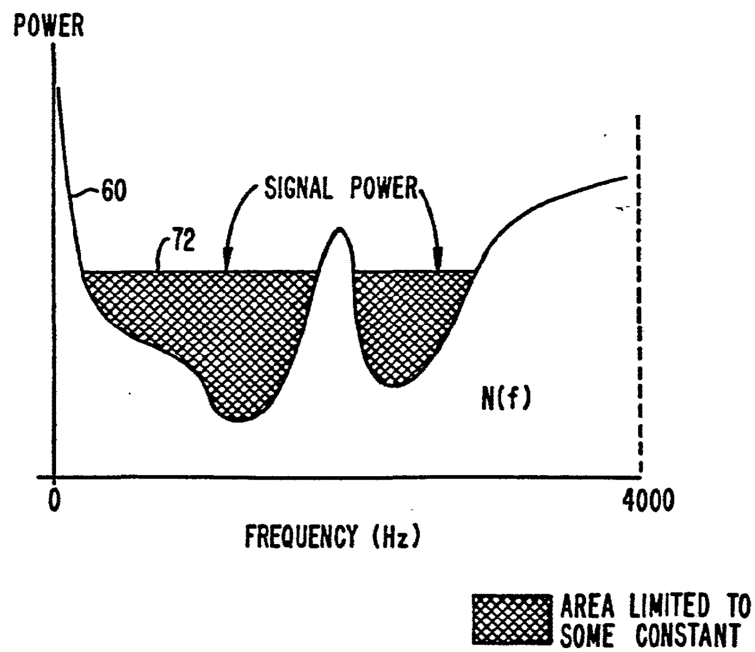

26
ORIGINATE
26'
ANSWER
(1)
GO OFF HOOK DIAL ANSWER MODEM
GO OFF HOOK
(2)
RECEIVE ANSWER EPOCH COMB COMPARE RECEIVED AMPLITUDES WITH TRANSMITTED AMPLITUDES
TRANSMIT ANSWER EPOCH COMB
(3)
ACCUMULATE NOISE DATA FFT ON NOISE DATA TO MEASURE AMPLITUDE OF NOISE COMPONENTS AT EACH CARRIER FREQ.
ACCUMULATE NOISE DATA FFT ON NOISE DATA TO MEASURE AMPLITUDE OF NOISE COMPONENTS AT EACH CARRIER FREQ.
(4)
TRANSMIT ORIGINATE COMB
RECEIVE ANSWER COMB COMPARE RECEIVED AMPLITUDES WITH TRANSMITTED AMPLITUDES
(5)
TRANSMIT FIRST PHASE-ENCODED SIGNAL
RECEIVE FIRST PHASE-ENCODED SIGNAL
(6)
RECEIVE SECOND PHASE-ENCODED SIGNAL
TRANSMIT SECOND PHASE-ENCODED SIGNAL
(7)
TRANSMIT DATA ON NUMBER OF BITS AND POWER LEVELS ON EACH CARRIER IN ANSWER-ORIGINATE DIRECTION
RECEIVE DATA
(8)
RECEIVE DATA
TRANSMIT DATA ON NUMBER OF BITS AND POWER LEVELS ON EACH CARRIER IN ORIGINATE-ANSWER DIRECTION

FIG._4.



POWER=1.00
E_b/N_o=0.5
POWER=10.0
E_b/N_o=2.50
(+3,+3)
0 BITS
2 BITS
4 BITS
POWER=20.00
E_b/N_o=4.00
(+3,+5)
(+5,+3)
5 BITS
POWER=32.75
E_b/N_o=5.46
(+7,+7)
6 BITS

FIG._5.



POWER
60
72
SIGNAL POWER
N(f)
0
4000
FREQUENCY (Hz)
AREA LIMITED TO
SOME CONSTANT

FIG._6.

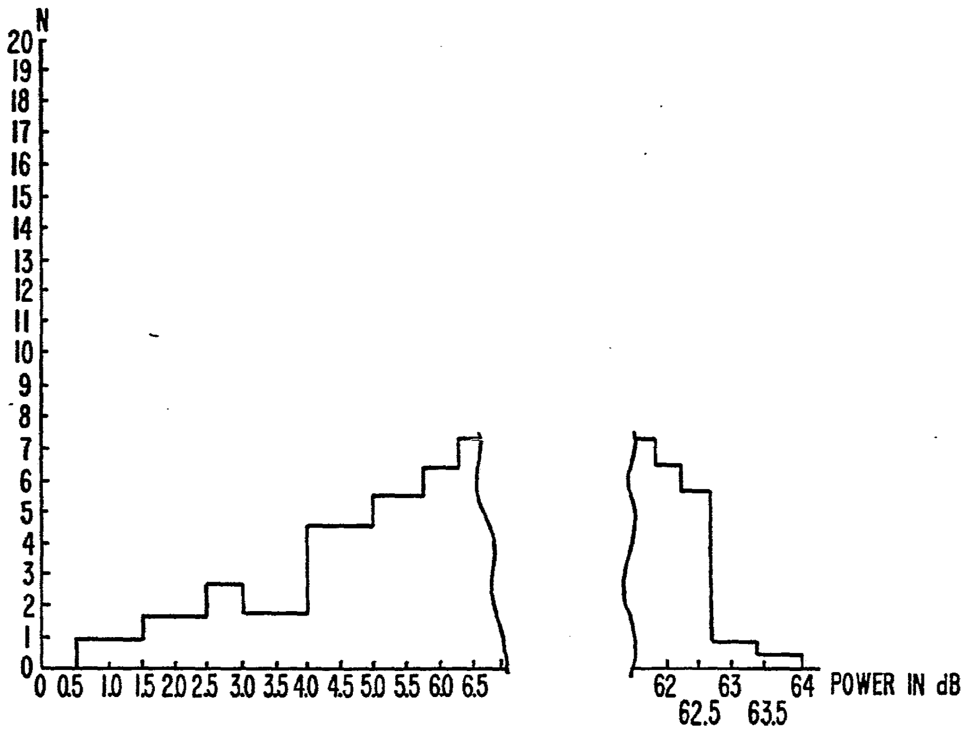

N
20
19
18
17
16
15
14
13
12
11
10
9
8
7
6
5
4
3
2
1
0
0 0.5 1.0 1.5 2.0 2.5 3.0 3.5 4.0 4.5 5.0 5.5 6.0 6.5
62 63 64 POWER IN dB
62.5 63.5

FIG._7.

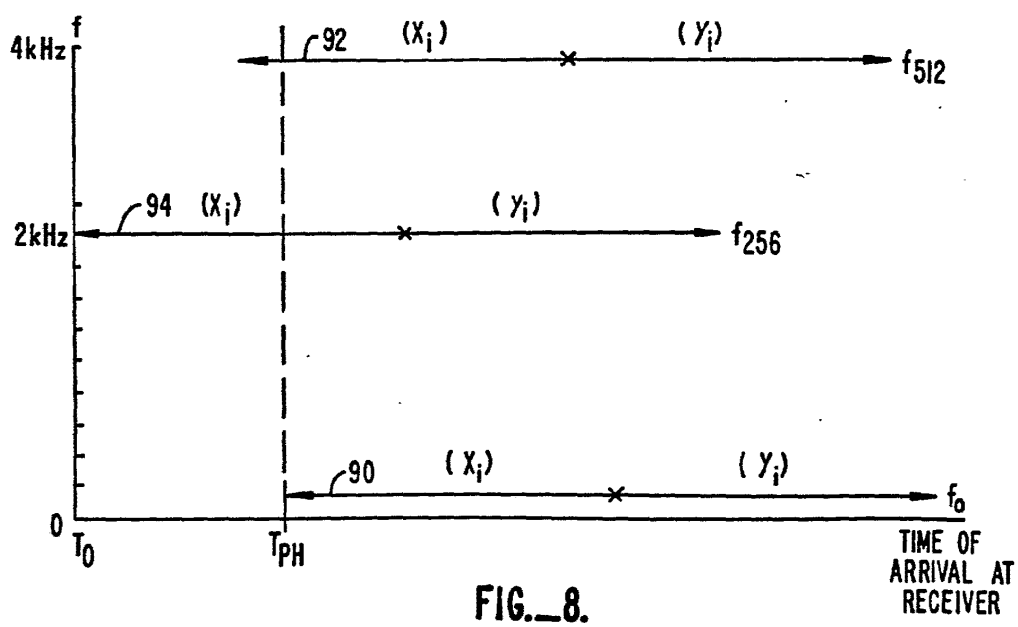

f
4kHz
2kHz
0
92
94
90
(X_i)
(Y_i)
f512
f256
f0
T0
TPH
TIME OF
ARRIVAL AT
RECEIVER

FIG._8.

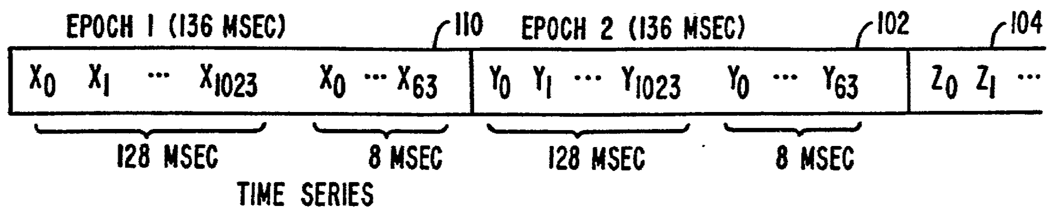


FIG._9.

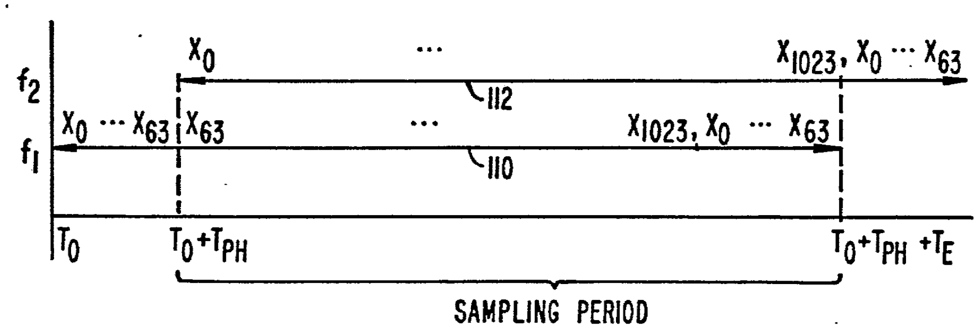


FIG._10.

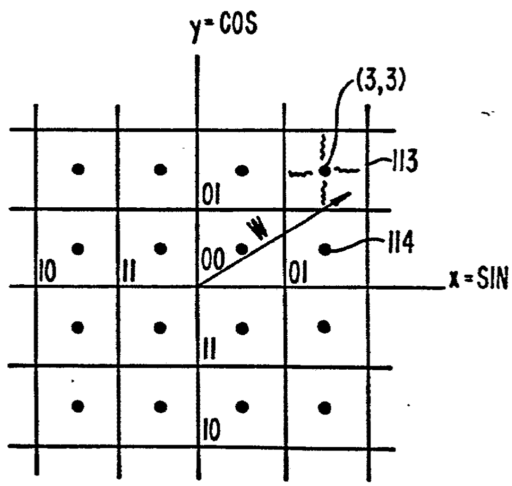


FIG._11.

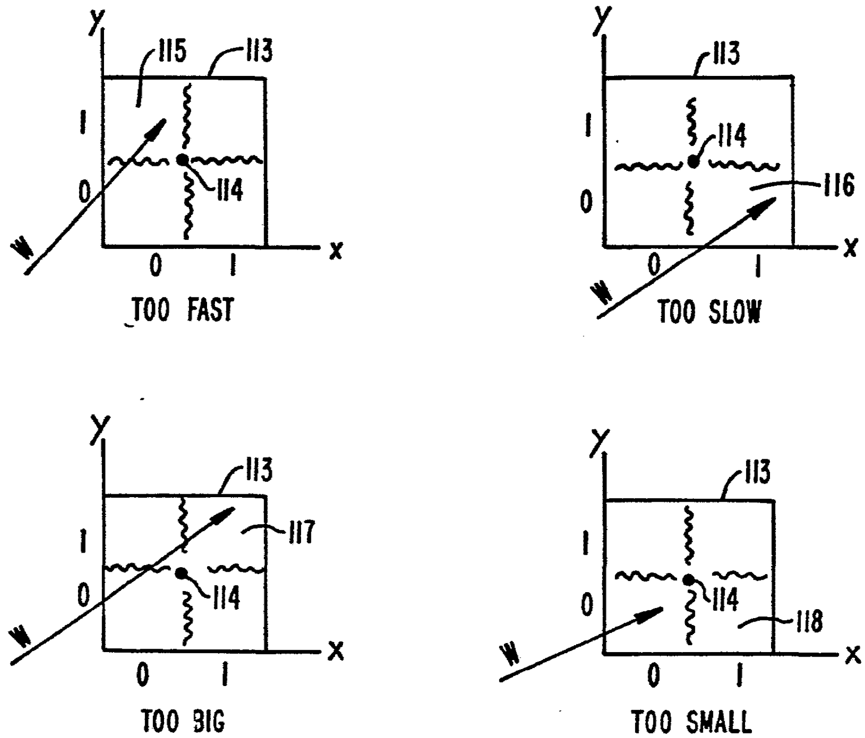

Y
115
113
1
0
114
W
0
1
X
TOO FAST
Y
113
1
114
0
116
W
0
1
X
TOO SLOW
Y
113
117
1
0
114
W
0
1
X
TOO BIG
Y
113
1
0
114
118
W
0
1
X
TOO SMALL

FIG._12.

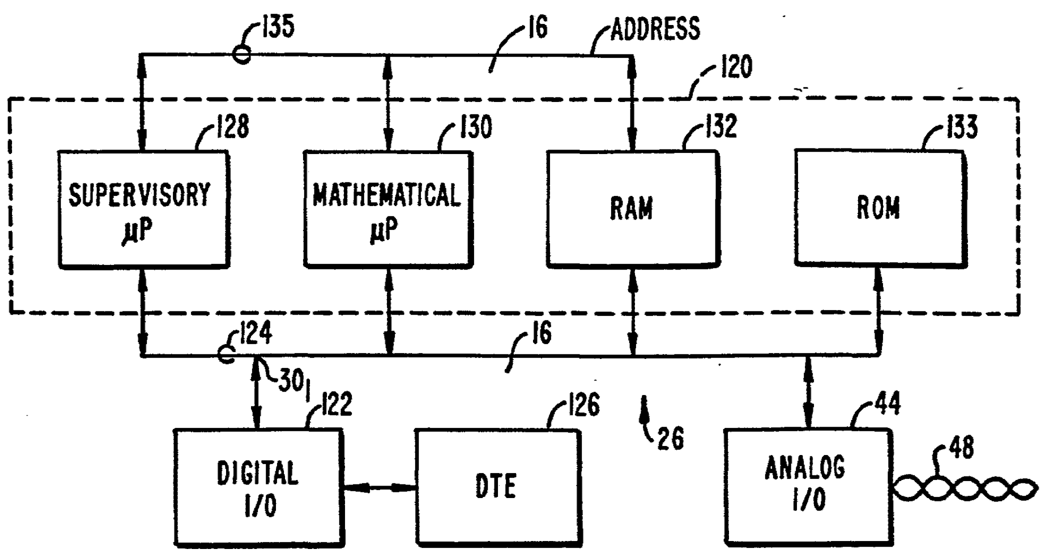

135
16
ADDRESS
120
128
130
132
133
SUPERVISORY μP
MATHEMATICAL μP
RAM
ROM
124
16
30
122
126
26
44
48
DIGITAL I/O
DTE
ANALOG I/O

FIG._13.

INTERNATIONAL SEARCH REPORT

International Application No PCT/US86/00983

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all) [3]

According to International Patent Classification (IPC) or to both National Classification and IPC

IPC (4): H04M 11/00;H04B 15/00,1/10;H04L 5/00,25/08;H04B 1/10
U.S. Cl.: 179/2DP; 375/39,58,99; 455/63

II. FIELDS SEARCHED

Minimum Documentation Searched [4]

Classification System	Classification Symbols
U.S.	179/2DP; 375/38,39,40,58,118; 370/16,108; 455/63,68+; 340/825.15

Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched [5]

III. DOCUMENTS CONSIDERED TO BE RELEVANT [14]

Category *	Citation of Document, [16] with indication, where appropriate, of the relevant passages [17]	Relevant to Claim No. [18]
X,P	Telecommincations, Volume 19, No. 10, issued October 1985 (Dedham, Massachusetts), H.R. Johnson, "PC Communications: The Revolution Is Coming", see pages 58j to 58r.	1-17
A	US, A, 4,438,511 (Baran) 20 March 1984	1-17
A,P	US, A, 4,559,520 (Johnston) 17 December 1985	1-17
A	US, A, 4,206,320 (Keasler et al.) 03 June 1980	1-17
A	US, A, 3,810,019 (Miller) 07 May 1974	1-5,10-12,17
A	US, A, 4,328,581 (Harmon et al.) 04 May 1982	1-5,10-12,17
A	US, A, 3,971,996 (Motley et al.) 27 July 1976	6-8,13-15
A,P	US, A, 4,555,790 (Betts et al.) 26 November 1985	6-8,13-15

(cont'd)

* Special categories of cited documents: [15]

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"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

IV. CERTIFICATION

Date of the Actual Completion of the International Search [2]	Date of Mailing of this International Search Report [2]
17 June 1986	10 JUL 1986
International Searching Authority [1]	**Signature of Authorized Officer [20]**
ISA/US	Matthew E. Connors Matthew E. Connors

Form PCT/ISA/210 (second sheet) (October 1981)

International Application No. PCT/US86/00983

III. DOCUMENTS CONSIDERED TO BE RELEVANT (CONTINUED FROM THE SECOND SHEET)

Category *	Citation of Document, [16] with indication, where appropriate, of the relevant passages [17]	Relevant to Claim No [18]
A	US, A, 3,783,385 (Dunn et al.) 01 January 1974	1-5
A	US, A, 4,047,153 (Thirion) 06 September 1977	1-5
A	US, A, 4,494,238 (Groth, Jr.) 15 January 1985	1-5
A	US, A, 4,495,619 (Acampora) 22 January 1985	1-5,10-12,17
A	US, A, 4,484,336 (Catchpole et al.) 20 November 1984	1-5,10-12,17
A	US, A, 4,459,701 (Lamiral et al.) 10 July 1984	9,16,17
A	US, A, 3,755,736 (Kaneko et al.) 28 August 1973	9,16,17
A	US, A, 4,315,319 (White) 09 February 1982	1-5,10-12,17
A,P	US, A, 4,573,133 (White) 25 February 1986	1-5,10-12,17
A	US, A, 4,392,225 (Wortman) 05 July 1983	1-5,10-12,17

Form PCT/ISA 210 (extra sheet) (October 1981)

PCT WELTORGANISATION FÜR GEISTIGES EIGENTUM
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INTERNATIONALE ANMELDUNG VERÖFFENTLICHT NACH DEM VERTRAG ÜBER DIE INTERNATIONALE ZUSAMMENARBEIT AUF DEM GEBIET DES PATENTWESENS (PCT)

(51) Internationale Patentklassifikation 6 : **H04L 5/14**	**A1**	**(11) Internationale Veröffentlichungsnummer:** **WO 97/01900** **(43) Internationales Veröffentlichungsdatum:** 16. Januar 1997 (16.01.97)

(21) Internationales Aktenzeichen: PCT/AT96/00112

(22) Internationales Anmeldedatum: 21. Juni 1996 (21.06.96)

(30) Prioritätsdaten:
A 1087/95 26. Juni 1995 (26.06.95) AT

(71) Anmelder *(für alle Bestimmungsstaaten ausser US)*: ERICSSON AUSTRIA AG [AT/AT]; Pottendorfer Strasse 25-27, A-1121 Wien (AT).

(72) Erfinder; und
(75) Erfinder/Anmelder *(nur für US)*: PFIEFFER, Johann [AT/AT]; Siedlungsstrasse 19, A-3804 Allentsteig (AT).

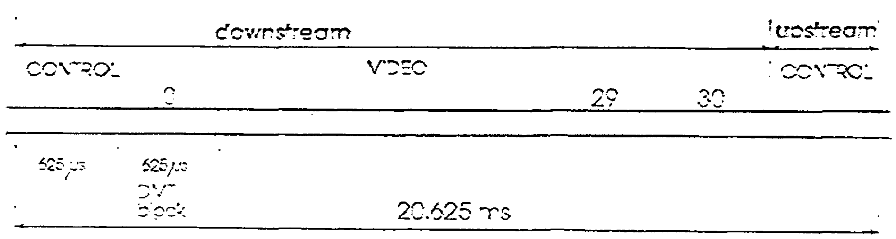
(74) Anwalt: GIBLER, Ferdinand; Dorotheergasse 7, A-1010 Wien (AT).

(81) Bestimmungsstaaten: AL, AM, AT, AU, AZ, BB, BG, BR, BY, CA, CH, CN, CZ, DE, DK, EE, ES, FI, GB, GE, HU, IL, IS, JP, KE, KG, KP, KR, KZ, LK, LR, LS, LT, LU, LV, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, TJ, TM, TR, TT, UA, UG, US, UZ, VN, ARIPO Patent (KE, LS, MW, SD, SZ, UG), eurasisches Patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), europäisches Patent (AT, BE, CH, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI Patent (BF, BJ, CF, CG, CI, CM, GA, GN, ML, MR, NE, SN, TD, TG).

Veröffentlicht
Mit internationalem Recherchenbericht.
Vor Ablauf der für Änderungen der Ansprüche zugelassenen Frist. Veröffentlichung wird wiederholt falls Änderungen eintreffen.

(54) Title: METHOD OF BI-DIRECTIONAL DATA TRANSMISSION OVER A TWO-WIRE LINE

(54) Bezeichnung: VERFAHREN ZUR BIDIREKTIONALEN DATENÜBERTRAGUNG ÜBER EINE ZWEIDRAHTLEITUNG



(57) Abstract

Proposed is a method of bi-directional data transmission over a two-wire line. Digital data destined for transmission or reception, e.g. using discrete multitone modulation (DMT), are modulated or demodulated as appropriate and separated by time-division multiplexing. The appropriate multiplex time frame is subdivided into a predetermined number N of time slots and a number K of those time slots is assigned exclusively to one direction, e.g. transmission, the remaining time slots (= N-K in number) being assigned exclusively to the other direction (e.g. reception).

(57) Zusammenfassung

Verfahren zur bidirektionalen Datenübertragung über eine Zweidrahtleitung, wobei digitale Daten zum Senden oder Empfangen, z.B. mittels diskreter Mehrtonmodulation (DMT), moduliert bzw. demoduliert und die zu sendenden und zu empfangenden Daten durch Zeitmultiplexbetrieb getrennt werden, wobei der zugehörige Multiplex-Zeitrahmen in eine vorbestimmbare Anzahl N von Zeitschlitzen unterteilt wird, und davon eine Anzahl K von Zeitschlitzen ausschließlich einer Übertragungsrichtung, z.B. Senden, und die restliche Anzahl (N-K) von Zeitschlitzen ausschließlich der anderen Übertragungsrichtung, z.B. Empfangen, zugeordnet wird.

LEDIGLICH ZUR INFORMATION

Codes zur Identifizierung von PCT-Vertragsstaaten auf den Kopfbögen der Schriften, die internationale Anmeldungen gemäss dem PCT veröffentlichen.

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FR	Frankreich	**MR**	Mauretanien	**VN**	Vietnam
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Verfahren zur bidirektionalen Datenübertragung über eine Zweidrahtleitung

Die Erfindung betrifft ein Verfahren zur bidirektionalen Datenübertragung über eine Zweidrahtleitung, wobei digitale Daten zum Senden oder Empfangen, z.B. mittels diskreter Mehrtonmodulation (DMT), moduliert bzw. demoduliert und die zu sendenden und zu empfangenden Daten, z.B. durch Frequenzmultiplexbetrieb (FDM) oder Echoauslöschung (EC), getrennt werden.

Um störende Beeinflussung von zu übermitteltenden Daten zu beseitigen, führen bekannte Verfahren dieser Art die Trennung der z.B. DMT-modulierten Daten im Frequenzmultiplexbetrieb (FDM) durch, wobei unterschiedliche Frequenzbereiche für die beiden Übertragungsrichtungen festgelegt sind. Eine weitere Möglichkeit zur Trennung besteht in der Anwendung des Echoauslöschungsverfahrens (EC), bei dem durch den Einsatz adaptiver Filter der Einfluß des Sendeteils auf den Empfänger durch adaptive Filter unterdrückt wird. Andere Trennverfahren wurden im Stand der Technik in diesem Zusammenhang bisher nicht verwendet.

Das FDM-Verfahren erzeugt bei der Übertragung entsprechend den beiden Übertragungsrichtungen ein unteres und ein oberes Frequenzband. Da aber die Kabeldämpfung frequenzabhängig ist, bereitet es große Schwierigkeiten für beide Übertragungskanäle die gleiche Übertragungsqualität zu erzielen, in den überwiegenden Fällen ist die Übertragungsqualität in eine besser als in die andere Richtung. Generell ist es aber erwünscht, eine möglichst gleiche Qualität für beide Kanäle anbieten zu können. Weiters ist bei FDM die Variation der Übertragungskapazität mit erheblichem Aufwand verbunden, da dafür eine Anpassung der jeweils verwendeten Bandfilter erforderlich ist, sodaß die Kanalbandbreite entsprechend erhöht oder erniedrigt werden kann.

Das weiters aus dem Stand der Technik bekannte Echoauslöschungs-Verfahren weist ebenso wenn auch anders geartete Nachteile auf. So ist bei diesem Verfahren das Nah-Nebensprechen ein großes technisches Problem, da der Signalabstand zwischen Sende- und Empfangssignal sehr groß ist. Es müssen daher sehr hohe Anforderungen an die bei den Sende- und Empfangsteilen vorgesehenen A/D-Wandler erfüllt werden, da Sende- und Empfangssignale gleichzeitig auftreten und diese entprechend gut getrennt werden müssen. Die hohen Pegelunterschiede der Sende- und Empfangssignale erfordern eine dementsprechend hohe Auflösung der A/D-Wandler, die wiederum höhere Produktkosten zur Folge hat.

Für die Durchführung dieser bekannten Trennmethoden FDM und Echoauslöschung ist auch eine relativ hohe Rechnerleistung erforderlich, die die Kosten für die Datenübertragung stark erhöhen. Besonders bei Anwendung in Fällen, in denen wie etwa bei ADSL (Asymmetric Digtal Subscriber Line) in einer Übertragungsrichtung ("downstream) große Datenraten von einer zentralen Datenanlage zu einem an der Peripherie gelegenen Teilnehmer und vergleichsweise geringe Datenraten in die andere Übertragungsrichtung ("upstream")

übermittelt werden sollen, ist der bei diesen bekannten Datenübertragungsverfahren getriebene Aufwand nur einer schlechten Nutzung unterworfen.
Ziel der Erfindung ist es, ein Verfahren anzugeben, das sich durch geringe Komplexheit hinsichtlich Hardware-Einsatz bzw. Rechnerleistung auszeichnet, sodaß die Durchführung auf einfache und kostengünstige Weise erfolgen kann.
Weiters ist es Ziel der Erfindung, ein Verfahren zu schaffen, mit dem sich bei Übertragungen, die zu einem großen Teil nur in einer der beiden Übertragungsrichtungen vor sich gehen, mit hoher Übertragungsgeschwindigkeit durchführen lassen.
Weitere Aufgabe der Erfindung ist es, eine sehr gute Übertragungsqualität mit relativ geringem technischen Aufwand zu erreichen, wobei eine Änderung der Übertragungskapazität einfach und kostengünstig möglich sein soll.
Erfindungsgemäß wird dies dadurch erreicht, daß die zu sendenden und zu empfangenden Daten durch Zeitmultiplexbetrieb (TDM) getrennt werden, wobei der zugehörige Multiplex-Zeitrahmen in eine vorbestimmbare Anzahl N von Zeitschlitzen unterteilt wird, und davon eine Anzahl K von Zeitschlitzen ausschließlich einer Übertragungsrichtung, z.B. Senden, und die restliche Anzahl (N-K) von Zeitschlitzen ausschließlich der anderen Übertragungsrichtung, z.B. Empfangen, zugeordnet wird.
Da beim erfindungsgemäßen Verfahren entweder nur Sender- oder nur Empfängerfunktionen aktiv sind, wird weniger Prozessorleistung als bei herkömmlichen Verfahren benötigt, da letztere einen sehr hohen internen Datenverkehr zu bewältigen haben. Dadurch gelingt es, eine nach dem erfindungsgemäßen Verfahren durchgeführte Übertragung sehr kostengünstig zu implementieren.
Das erfindungsgemäße Verfahren bietet weiters den Vorteil einer gleichen Übertragungsqualität in beiden Übertragungsrichtungen, da Senden und Empfangen bei TDM mit der gleichen Leitungsdämpfung erfolgt. Dadurch können beide Übertragungsrichtungen mit geringstmöglicher Qualtitätsminderung im gleichen Frequenzbereich durchgeführt werden. Ein weiterer Vorteil des erfindungsgemäßen Verfahrens ist die sehr einfache Veränderung der Übertragungskapazität, die durch die entspechende Wahl der Anzahl der Zeitschlitze für die jeweilige Übertragungsrichtung ermöglicht wird.
Als besonders vorteilhaft bei asymmetrischer Datenübertragung kann es sein, wenn in einer Übertragungsrichtung der Großteil der Daten und in der anderen nur ein kleiner Rest übertragen wird. Dies ist dann gegeben, wenn die Anzahl N der Zeitschlitze sehr viel größer als die Anzahl K gewählt wird. Vorzugsweise ist diese Bedingung erfüllt, wenn N gleich 30 und K gleich 1 ist.
Da das erfindungsgemäße Verfahren zur Datenübertragung über Telephonleitungen eingesetzt werden kann, kann es z.B. durch die Nummernwahl auf der Leitung zu impulsartigen Störungen kommen, die einen Übertragungsfehler bewirken, der unbedingt korrigiert werden muß. Die Datenübertragung muß aber nicht über Telephonleitungen erfolgen, sie kann im Rahmen der Erfindung über jede dafür geeignete Zweidrahtleitung

geschehen. Genauso können die unterschiedlichsten elektromagnetischen Störungen, auch systemexterne, auf die Datenübertragung ihren Einfluß haben.
Das bekannte ARQ (Automatic Repeat Request)-Verfahren wird zur Fehlerkorrektur üblicherweise so eingesetzt, daß die Datenübertragung auch bei beliebigen Störungen auf der Leitung fehlerfrei bleibt, wobei der Datendurchsatz jedoch stark absinken kann, da ein fehlerhaft übertragenes Datenpaket solange wiederholt wird, bis es fehlerfrei empfangen wird.
In weiterer Ausbildung der Erfindung kann daher vorgesehen sein, daß im Multiplex-Zeitrahmen der Datenübertragung im Zeitmittel eine vorbestimmbare Anzahl von Zeitschlitzen für ARQ (Automatic Repeat Request)-Übertragungswiederholungen vorgesehen sind.
Bei dieser Ausführungsform steht somit ständig Übertragungs-Überkapazität zur Verfügung. Wird ein Datenblock fehlerhaft empfangen, fordert der Empfänger nur so oft eine Wiederholung an, wie es im Rahmen der im Zeitmittel zur Verfügung stehenden Überkapazität möglich ist, sodaß unbeeinflußt durch die Übertragungswiederholungen der nominelle Datendurchsatz konstant gehalten werden kann. Im fehlerfreien Übertragungsfall wird ein höher redundantes Signal übermittelt. Die Dauer der Zeitspanne, über die die Zeitmittelung erfolgt, ist im wesentlichen durch die Speicherkapazität des eingesetzten ARQ-Puffers begrenzt.
Nach einer anderen Variante der Erfindung kann vorgesehen sein, daß bei fehlerhafter Übertragung die Daten, z.B. mittels eines Rechenalgorithmus, modifiziert übertragen werden.
Dadurch kann der bei der Übertragung auftretende Fehler, der durch das Abschneiden eines Teils der Amplitude bei Sende-Übersteuerung hervorgerufen wird, korrigiert werden.
In besonders bevorzugter Weise kann dabei vorgesehen sein, daß die Daten durch logische Inversion modifiziert werden.
Diese Inversionsoperation stellt einen sehr einfach berechenbaren Algorithmus dar, der ohne großen Aufwand realisierbar ist.
Weiters kann vorgesehen sein, daß die Schaltfrequenz einer Störquelle, z.B. ein Netzteil, mit einer der Trägerfrequenzen der diskreten Mehrtonmodulation synchronisiert wird.
Dadurch kann das auf frequenzselektive Störungen empfindliche DMT-Verfahren gegen bekannte Störquellen gesichert werden. Bei Synchronisation der Schaltfrequenz der Störquelle auf eine der Trägerfrequenzen der DMT-Modulation wirkt sich die Störung nur auf diese Trägerfrequenz und deren Vielfache aus, sodaß sie durch einen adaptiven Algorithmus kompensiert werden können.
Bei mehreren nebeneinander geführten Zweidrahtleitungen, auf denen jeweils Daten übertragen werden, ergibt sich üblicherweise ein Übersprechen, welches auf die Übertragung naturgemäß störend wirkt.
Gemäß einer anderen Ausführungsform des erfindungsgemäßen Verfahrens, bei welchem Daten über zwei oder mehr Zweidrahtleitungen, die zumindest teilweise in

Übersprechabstand geführt sind, übertragen werden, kann vorgesehen sein, daß der Zeitmultiplex-Betrieb (TDM) auf allen Zweidrahtleitungen synchron durchgeführt wird, sodaß auf allen Zweidrahtleitungen gleichzeitig entweder gesendet oder empfangen wird.
Dadurch wird immer zur gleichen Zeit entweder gesendet oder empfangen, sodaß eine störende Beeinflussung der einzelnen Empfänger durch nicht direkt verbundene Sender vermieden werden kann.
Im folgenden wird die Erfindung anhand eines in den Zeichnungen dargestellten Ausführungsbeispieles näher erläutert.
Es zeigt dabei:
Fig.1 ein Blockschaltbild zur Durchführung einer Ausführungsform des erfindungsgemäßen Verfahrens und
Fig.2 eine schematische Darstellung eines erfindungsgemäßen Zeitrahmens.
Eine bidirektionale Datenübertragung von digitalen Daten gemäß dem in Fig.1 dargestellten Blockschaltbild wird durchgeführt, indem beim Senden die aus einer Datenquelle 1,4 kommenden digitalen Daten im Sendeteil 50 zu einem analogen Sendesignal umgewandelt und über einen Leitungsübertrager 13 einer Zweidrahtleitung 100 an einen am Ende dieser Leitung 100 gelegenen Teilnehmer übertragen werden. Demgegenüber wird ein auf der Zweidrahtleitung 100 ankommendes Signal über den Leitungsübertrager 13 als Empfangssignal an den Eingang eines Empfangteils 51 geführt und dort in digitale Daten umgewandelt. Da beim erfindungsgemäßen Verfahren nie gleichzeitig gesendet und empfangen wird, kann an Stelle einer sonst üblichen Gabelschaltung der Leitungsübertrager 13 verwendet werden, wodurch die oft problematische Anpassung der Gabelschaltung an die Leitungsimpedanz von vornherein wegfällt. Ein durch eine Gabelschaltung bedingtes störendes Übersprechen, durch welches Signalreste vom Sender zum Empfänger derselben Teilnehmerseite gelangen, scheidet somit als Störquelle für dieses Verfahren aus.
In dem in Fig.1 gezeigten Ausführungsbeispiel ist der Sende- und Empfangsteil 50, 51 sowohl einer zentralen Datenstelle C (CENTRAL) alsauch einer peripheren Datenstelle R (REMOTE) in einem einzigen Blockschaltbild dargestellt, welches so zu verstehen ist, daß die zentrale Datenstelle C über den Übertrager 13, die Zweidrahtleitung 100 und einen weiteren Übertrager 13 mit der Datenstelle R verbunden ist. Jene Funktionseinheiten, die nur zur Datenstelle C bzw. R zugehörig sind, sind mit "ATU-C only" bzw. "ATU-R only" gekennzeichnet.
Ohne Beschränkung der allgemeinen Anwendbarkeit des erfindungsgemäßen Verfahrens sei als Ausführungsbeispiel einer asymmetrischen Datenübertragung ein Heimvideosystem beschrieben, bei welchem in der zentralen Datenstelle C die Videoinformation verschiedener Videos in einem Großrechner als Daten in komprimierter Form gespeichert und über eine periphere Datenstelle R abrufbar ist. Über einen bidirektionalen Steuerkanal wird die Steuerinformation zwischen den Datenstellen C und R ausgetauscht, wobei eine Datenrate von 64 kbit/s festgelegt ist. Diese Steuerinformation kann sich auf verschiedene vom Teilnehmer auszugebende Befehle, wie etwa PLAY, REWIND o.ä, wie sie von einem

Videorecorder bekannt sind sowie interne Steuerkommandos beziehen und ist in ihrer Menge vergleichsweise gering gegenüber der von der zentralen Datenstelle C ausgesendeten Breitbandinformation, die im wesentlichen die Videoinformation beinhaltet, die mit einer Datenübertragungsrate von 2, 048 Mbit/s nur in einer Richtung von C zu R gesendet wird.

Die genannten Datenraten können jedoch für das erfindungsgemäße Verfahren aber auch gänzlich anders, z.B. viel höher gewählt werden, wobei für die nur in eine Richtung zu übermittelnde Breitbandinformation auch eine Datenrate von etwa 50 Mbit/s bis 150 Mbit/s zur Verfügung gestellt werden kann. Die übertragene Information kann dabei jede Art von Sprach-, Bild- oder Dateninformation darstellen. Ebenso ist eine andere Rate für den bidirektionalen Steuerkanal ausführbar, der aber nicht nur Steuerfunktionen sondern alle möglichen Datenübertragungsfunktionen erfüllen kann.

Am eingangsseitigen Teil des Sendeteils 50 sind für die Datenstelle C zwei verschiedene Dateneingänge und für die Datenstelle R nur ein Dateneingang ausgebildet. An den ersten Eingang, der für C und R gleich ist, gelangt der Datenstrom aus der Datenquelle 1, die z.B. im wesentlichen Steuerbefehle aussendet, die über einen nachfolgenden Verwürfler 2 in einen diesem nachfolgenden Sendepuffer 3 gelangen, wobei die aus der Datenquelle 1 kommenden Daten im Verwürfler 2 nach einem vorbestimmbaren Algorithmus gewandelt werden. Dadurch wird ein länger andauernder, konstanter logischer Zustand verhindert und eine ausgeglichene statistische Verteilung der binären Zustände erreicht. Anschließend daran erfolgt im Sendepuffer 3 eine Zwischenspeicherung der verwürfelten Signale. In der Datenstelle R sind die aus dem Sendepuffer 3 austretenden Daten über eine Vorrichtung MUX mit anderen Daten, die im ARQ-Puffer 24 erzeugt werden und Wiederholanweisungen enthalten, gemultiplext.

Am zweiten Eingang des Sendeteils 50, der nur für die Datenstelle C ausgeführt ist, kommt der Datenstrom aus der Datenquelle 4, die die Breitbandinformation generiert, über einen nachfolgenden Verwürfler 5 und über einen ARQ (Automatic Request)-Puffer 6, der einen CRC-Generator enthält, über den eine Fehlerkorrekturkodierung erfolgt, an den zweiten Eingang des Sendeteiles 50. Die im Verwürfler 5 umgewandelten Daten werden im ARQ-Puffer 6 zwischengespeichert und bei fehlerhafter Übertragung wiederholt. Eine besondere. erfindungsgemäße ARQ-Übertragungstechnik wird weiter unten noch beschrieben.

Die über die Eingänge des Sendeteils 50 seriell eintreffenden Daten werden im Kodierer 7 zum Herabsetzen der Datenrate in vorbestimmbarer Länge zusammengefaßt und anhand einer Kodiertabelle einem entsprechenden Symbol zur weiteren Verarbeitung zugeordnet. Weiters wird dieses kodierte Signal in dem nachfolgenden DMT (Discrete Multi Tone)-Modulator 8 nach diesem bekannten Verfahren moduliert und über ein Hochpaß-Filter 9 geleitet, welches zur Vermeidung von Störeinflüssen im wesentlichen das Sprachfrequenzband unterdrückt. Das digitale Ausgangssignal dieses Hochpaß-Filters 9 wird über einen Digital-Analog-Wandler 10 in ein analoges Signal gewandelt, welches über ein Bandpaß-Filter 11 und anschließend über einen Verstärker 12 zum Wandler 13 gelangt. Das Bandpaß-Filter 11 erfüllt einerseits nochmals die Funktion des Hochpasses 11 und andererseits schneidet es die durch den Analog-Digital-Wandler 10 hervorgerufenen hochfrequenten Spannungsspitzen ab. Die Frequenz der Analog-Digital-Wandlung ist zur Erfüllung des Abtasttheorems so gewählt, daß für die höchsten vorkommenden Frequenzen mindestens zweimal eine Abtastung durch den Analog-Digital-Wandler 10 erfolgt.

Der Sendeteil 50 und der Empfangsteil 51 sind durch eine TDM (Time Division Multiplex)-Einheit 30 gesteuert, sodaß erfindungsgemäß die zu sendenden und die zu empfangenden Daten durch Zeitmultiplexbetrieb getrennt werden, wobei der zugehörige Multiplex-Zeitrahmen in eine vorbestimmbare Anzahl N von Zeitschlitzen unterteilt wird, und davon eine Anzahl K von Zeitschlitzen des Zeitrahmens auschließlich einer Übertragungsrichtung, z.B. Senden, und die restliche Anzahl N-K von Zeitschlitzen ausschließlich der anderen Übertragungsrichutng, z.B. Empfangen, zugeordnet wird. Dazu steuert die TDM-Einheit den Sendeteil 50 und den Empfangsteil 51, indem sie zur gegebenen Zeit diese aktiviert. Der Sendeteil 50 und der Empfangsteil 51 sind dabei nie gleichzeitig in Betrieb, wodurch die für die Steuerung benötigte Prozessorleistung entsprechend niedrig ausgelegt werden kann. Da dadurch auch eine Beeinflussung des eigenen Senders auf den Empfänger ausgeschlosssen ist, ist für den Analog-Digital-Wandler 16 des Empfängerteils nur eine geringe Auflösung erforderlich. Dieser Vorteil wirkt sich infolge der direkten Proportionalität von Auflösung und Preis bei Analog-Digital-Wandlern sehr kostengünstig aus.

Das erfindungsgemäße Verfahren hat den Vorteil eines relativ geringen Bandbreitenbedarfes und einer sehr geringen Komplexheit, die sich bei der Hardware bzw. bei der benötigten Rechnerleistung zeigt. Bei herkömmlichen Verfahren zur Trennung von Senden und Empfangen geht ein beträchtlicher Teil der Rechnerleistung für interne Kommunikation verloren, während beim erfindungsgemäßen Verfahrern diese Rechner-Hilfskapazität sehr gering gehalten werden kann.

Das erfindungsgemäße Verfahren hat dort seine Grenze, wo sich der Anteil des Sendens und Empfangens der 50%-Prozentgrenze nähert, da dann andere Verfahren etwa wie Echo-Cancelling o.ä. mit gleichgroßem oder kleinerem Aufwand durchgeführt werden können.

In Fig.2 ist der in Zeitschlitze unterteilte Zeitrahmen, wie er im erfindungsgemäßen Verfahren zur Anwendung gelangt, dargestellt. Die beiden Übertragungsrichtungen sind durch die Ausdrücke "upstream" und "downstream" gekennzeichnet. Der ganze Zeitrahmen ist in diesem Beispiel 20,625 ms lang und in verschiedene Schlitze zu 625 μs aufgeteilt, wobei die Mehrzahl der Daten in downstream-Richtung übertragen wird. Diese Aufteilung ist besonders dann von Vorteil, wenn in einer Übertragungsrichtung ein bidirektionaler Kanal mit geringer und ein unidirektionaler Kanal mit hoher Datenrate benötigt wird. In dem dargestellten Ausführungsbeispiel werden über den bidirektionalen Kanal durch die mit CONTROL bezeichneten Zeitschlitze in downstream- und upstream-Richtung Steuerbefehle und über den unidirektionalen Kanal durch die mit VIDEO bezeichneten 30 downstream-Zeitschlitze mit im Zeitmittel einem Hilfsschlitz Videoinformation übertragen. Diese Art der Übertragung kann für beliebige Informationen erfolgen.

Die Verteilung der Sende- bzw. Empfangskapazitäten ist den jeweiligen Verhältnissen durch Wahl der Anzahl der upstream bzw. downstream-Zeitschlitze anpaßbar. Bei sich ändernden Auslastungen kann dieses Verhältnis automatisch entsprechend dem aktuellen Bedarf abgestimmt werden. Die festgelegten Sende- und Empfangszeiten haben gegenüber einer Frequenzmultiplex-Übertragung den Vorteil, daß nicht gleichzeitig empfangene und zu sendende Daten verarbeitet werden müssen, wodurch die Rechnerleistung bzw. der Hardware-Aufwand entsprechend niedrig ausgelegt werden kann. In jedem DMT-Schlitz wird eine codierte und DMT-modulierte Dateneinheit übertragen.

Für ARQ-Übertragungswiederholungen wird gemäß einer erfindungsgemäßen Ausführungsform im Multiplex-Zeitrahmen der Datenübertragung im Zeitmittel eine vorbestimmbare Anzahl von Zeitschlitzen für ARQ-Übertragungswiederholungen vorgesehen sind. Dazu werden beim Senden der Daten diese ständig in den ARQ-Sendepuffer 6 eingeschrieben und von diesem wieder an den Kodierer 7 weitergegeben. Dabei werden die vom Puffer 6 abgehenden Daten schneller übertragen als dieser gefüllt wird. In der dabei entstehenden Lücke wird erneut jeweils der letzte Datenblock eingetragen, dieser wird jedoch empfängerseitig als wiederholter Block erkannt und automatisch beseitigt. Somit wird im fehlerfreien Übertragungsfall ständig mit Überkapazität gesendet, ohne daß der übertragene Informationsgehalt größer ist.

Sobald ein Übertragungsfehler auftritt, erkennt der Empfänger in der peripheren Datenstelle R den Fehler mittels seiner CRC- Fehlererkennung in der ARQ-Einheit 24 und gibt darauf den Befehl über den Multiplexer des Sendepuffers 3 zur Datenwiederholung weiter, der dann als Steuerinformation über den bidirektionalen Kanal gesendet wird. In der zentralen Datenstelle C wird diese Information nach Durchlaufen des Empfängerteils 51 im Empfängerpuffer 27 gedemultiplext und ein Steuerbefehl an den ARQ-Puffer 6 gegeben, die fehlerhafte Übertragung zu wiederholen.

Dafür steht in diesem Ausführungsbeispiel im Zeitmittel nur ein Hilfsschlitz zur Verfügung, was einer Überkapazität von 3,33% entspricht. Dauer und Anzahl der Hilfsschlitze sind in diesem Zusammenhang keiner Einschränkung unterworfen und können innerhalb des technisch Realisierbaren beliebig den jeweiligen Verhältnissen angepaßt werden.

Nach einer Fehlübertragung wird nun im darauffolgenden Zeitrahmen, die Wiederholungsübertragung durchgeführt, die sich über mehrere nacheinanderfolgende Zeitschlitze erstrecken kann. Gemittelt über die Zeit sollte in diesem Beispiel nur ein Zeitschlitz pro Rahmen für die Wiederholungen benutzt werden.

Die Zeitspanne, über die dabei das Zeitmittel berechnet wird, ist durch die Größe des ARQ-Pufferspeichers festgelegt. Sobald dieser mit Information vollgeschrieben ist, können keine weiteren Wiederholungen durchgeführt werden und der fehlerhafte Datenblock muß als transparent ausgegeben werden.

Gegenüber einem herkömmlichen ARQ-Verfahren ist die für die Datenwiederholungen festgelegte Zeitspanne im Zeitmittel fixiert. Dadurch kann es nicht passieren, daß aufgrund einer länger andauernden Störung die Übertragung solange wiederholt wird bis sie fehlerfrei ist und damit die Übertragungszeit sich stark erhöht. Durch das bekannte ARQ-Verfahren wird die Datenübertragung auch bei beliebigen Störungen solange wiederholt, bis sie fehlerfrei empfangen wird, wodurch der Datendurchsatz aber sehr stark sinkt. Hingegen wird durch die feste Überkapazität, die zwischen 2 und 10%, vorzugsweise aber zwischen 3 und 5% liegt, im erfindungsgemäßen Verfahren die Übertragung nur so oft wiederholt, wie es im Rahmen der Überkapazität möglich ist, um den nominellen Datendurchsatz aufrecht zu erhalten. Kann bei mehreren aufeinanderfolgenden falschen Datenblöcken einer nicht mehr wiederholt und richtig empfangen werden, wird er transparent ausgegeben.

Bei einem durch die diskrete Mehrtonmodulation (DMT) modulierten Signal ist das Verhältnis von Spitzenwert zu Mittelwert sehr groß, sodaß ein Abkappen ("Clipping") der Signalspitze eine häufige Fehlerquelle darstellt. Um diesen Fehler auf einfache Weise zu korrigieren, kann nach einer fehlerhaften Datenübertragung die digitale Bitfolge beim Wiederholvorgang im Sender z.B. durch einen Rechenalgorithmus, modifiziert werden und dann erneut übertragen werden. Im Empfänger wird der verwendete Rechenalgorithmus entsprechend in Umkehrung angewendet und die Daten wiedergewonnen. Dadurch kann dieser Übertragungsfehler sehr effektiv ausgeschaltet werden. Im besonderen ist es schaltungs- oder rechentechnisch auf einfache Weise durchführbar, die fehlerhaften Daten in invertierter Form zu übertragen

Eine weitere Störquelle beim DMT-Verfahren ergibt sich aus der Schaltfrequenz der eingesetzten Spannungsversorgung, z.B. des Netzteils, da diese Schaltfrequenz im Übertragungsbereich liegt und somit als frequenzselektive Störung ihre Auswirkung zeigt. Hinzu kommt die Abhängigkeit dieser Störungen von anderen Einflußgrößen, etwa die gerade am Netzteil vorliegende Last. Diese Art von Störungen können verringert werden, indem die Schaltfrequenz des Netzteils auf eine der Trägerfrequenzen der DMT-Modulation synchronisiert wird. Damit wirkt sich diese Störung nur auf diese Trägerfrequenz und ihre Vielfache aus, sodaß sie sehr leicht durch einen adaptiven Algorithmus kompensiert werden können.

In Fig.1 ist weiters der dem Sendeteil 50 entsprechende Empfangsteil 51 dargestellt. Die über die Zweidrahtleitung 100 und den Übertrager 13 von der anderen Teilnehmerseite einlangenden Signale werden über einen Bandpaß 14 und über eine AGC (Automatic Gain Control)-Einheit, die unabhängig von den momentanen Signalverhältnissen auf der Leitung ein annähernd amplitudenkonstantes Signal erzeugt, an den Eingang eines zum Empfangsteil 51 gehörigen Analog-Digital-Wandlers 16 geführt, dessen Ausgang mit einem Hochpaß-Filter 17 verbunden ist. Das am Eingang des Hochpasses 17 anliegende Signal wird über einen AGC-Regelkreis 18 als Stellgröße zur AGC-Einheit 15 rückgeführt.

Nach dem Hochpaß 17 erfolgt die Demodulation des Signals, aus welchem nur in der peripheren Datenstelle R der mitübertragene Pilotton einer Pilot-AGC-Einheit 20 zugeführt wird, woraus in der Taktgewinnungseinheit 21 ein Referenzsignal für die Takterzeugungseinheit 31 der peripheren Datenstelle R gewonnen wird. Diese Takterzeugungseinheit 31 generiert für die TDM-Einheit 30 und für den Systemtakt die

Zeitbasis. Die Datenstelle C benötigt keine Taktgewinnungseinheit, da hier eine unabhängige Zeitbasis vorgesehen ist.
Die durch die Übertragungsstrecke bewirkten linearen Verzerrungen werden in einem an den DMT-Demodulator 19 anschließenden Entzerrer 22 mit update-Funktion beseitigt. Daran anschließend findet in einem Dekodierer 23 das Umschlüsseln entsprechend einer Dekodiertabelle statt, woraufhin am Ausgang des Dekodieres 23 wieder ein serieller Bitstrom vorliegt, der über zwei Ausgänge geführt wird. Der für Datenstelle C und R gleich ausgeführte erste Ausgang besteht aus einem Empfangs-Puffer 27 für Steuerinformation, einem nachfolgenden Entwürfler 28, in welchem die Daten in ihrer richtigen Reihenfolge wiederhergestellt werden und der Datensenke 29, die die gesendeten Steuerdaten empfängt. Der zweite Ausgang des Empfangsteils 51, welcher nur für die Datenstelle R vorgesehen ist, ist mit einem ARQ-Puffer 24 verbunden, der die übertragene Breitbandinformation aus der Datenstelle C zwischenspeichert, verifiziert und bei Bedarf über eine im ARQ-Puffer 24 integrierte Steuereinheit den Befehl zum nochmaligen Senden der fehlerhaft übertragenen Daten an den Multiplex-Eingang des Sendepuffers 3 gibt, der zur Datenstelle C rückübertragen wird. Am Ausgang des ARQ-Puffers 24 ist ein Entwürfler 25 und daran anschließend eine Datensenke 26 zur Übernahme der Breitbandinformation angeschlossen.
Werden Daten über zwei oder mehr Zweidrahtleitungen, die zumindest teilweise in Übersprechabstand geführt sind, übertragen, kann es geschehen, daß durch die gegenseitige induktive Beeinflussung der Zweidrahtleitungen es zum Übersprechen kommt. Besonders in einer zentralen Datenanlage, in der viele abgehende Zweidrahtleitungen nebeneinander geführt werden, kann es zu dieser unerwünschten Störung kommen.
Bei einer Ausführungsform des erfindungsgemäßen Verfahrens wird diese Art der Störung vermieden, indem der Zeitmultiplex-Betrieb auf allen Zweidrahtleitungen synchron durchgeführt wird. Dies bedeutet, daß gleichzeitig über alle Zweidrahtleitungen entweder gesendet oder empfangen wird, sodaß keine Beeinflussung mehr möglich ist.

Patentansprüche

1. Verfahren zur bidirektionalen Datenübertragung über eine Zweidrahtleitung, wobei digitale Daten zum Senden oder Empfangen, z.B. mittels diskreter Mehrtonmodulation (DMT), moduliert bzw. demoduliert und die zu sendenden und zu empfangenden Daten, z.B. durch Frequenzmultiplexbetrieb (FDM) oder Echoauslöschung (EC), getrennt werden, **dadurch gekennzeichnet**, daß die zu sendenden und zu empfangenden Daten durch Zeitmultiplexbetrieb (TDM) getrennt werden, wobei der zugehörige Multiplex-Zeitrahmen in eine vorbestimmbare Anzahl N von Zeitschlitzen unterteilt wird, und davon eine Anzahl K von Zeitschlitzen ausschließlich einer Übertragungsrichtung, z.B. Senden, und die restliche Anzahl (N-K) von Zeitschlitzen ausschließlich der anderen Übertragungsrichtung, z.B. Empfangen, zugeordnet wird.

2. Verfahren nach Anspruch 1, **dadurch gekennzeichnet**, daß N gleich 30 und K gleich 1 ist.

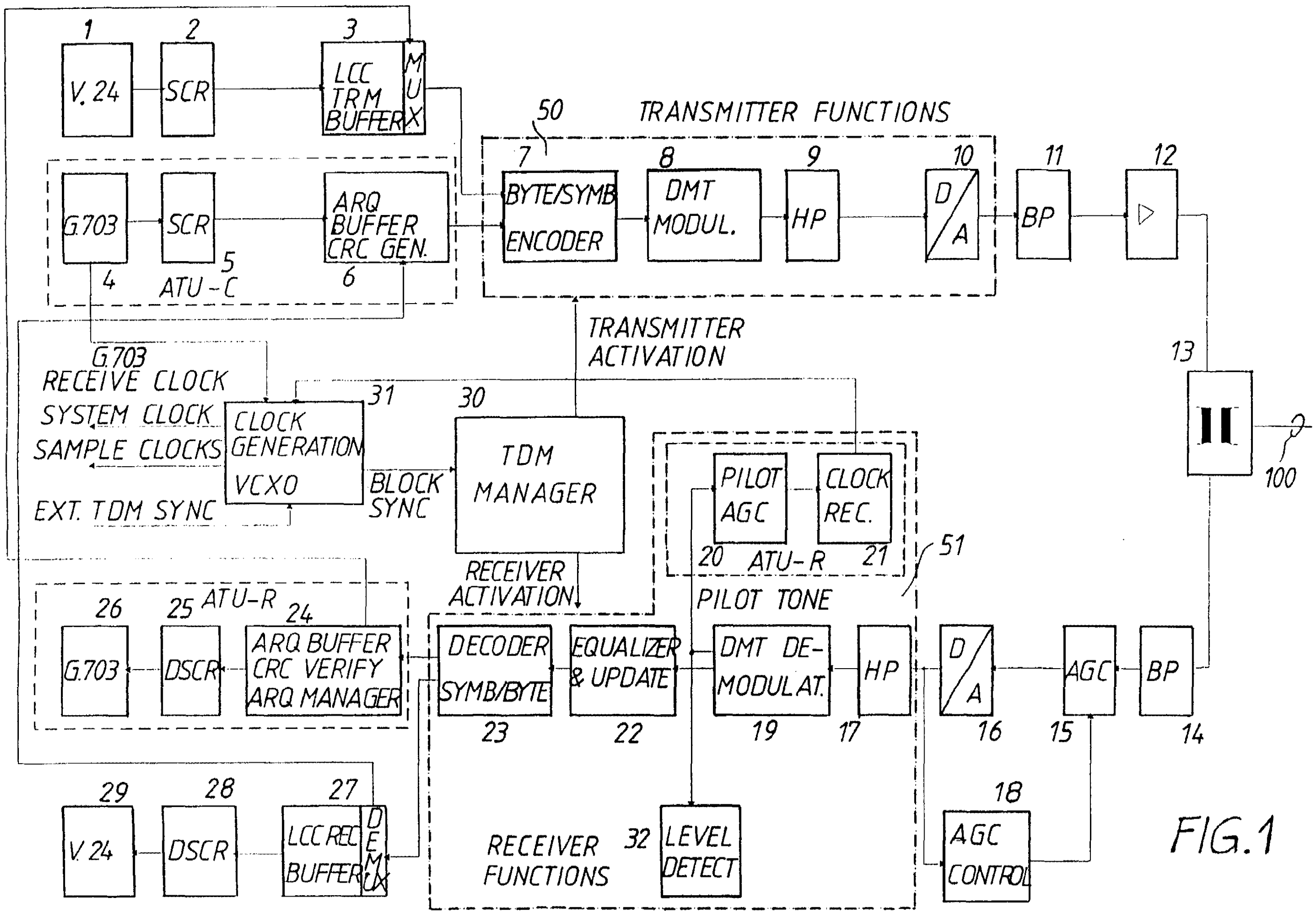
3. Verfahren nach Anspruch 1 oder 2, **dadurch gekennzeichnet**, daß im Multiplex-Zeitrahmen der Datenübertragung im Zeitmittel eine vorbestimmbare Anzahl von Zeitschlitzen für ARQ (Automatic Repeat Request)-Übertragungswiederholungen vorgesehen sind.

4. Verfahren nach Anspruch 1, 2 oder 3, **dadurch gekennzeichnet**, daß bei fehlerhafter Übertragung die Daten, z.B. mittels eines Rechenalgorithmus, modifiziert wiederholt übertragen werden.

5. Verfahren nach Anspruch 4, **dadurch gekennzeichnet**, daß die Daten durch logische Inversion modifiziert werden.

6. Verfahren nach Anspruch 1 bis 5, **dadurch gekennzeichnet**, daß die Schaltfrequenz einer Störquelle, z.B. ein Netzteil, mit einer der Trägerfrequenzen der diskreten Mehrtonmodulation synchronisiert wird.

7. Verfahren nach Anspruch 1 bis 6, wobei Daten über zwei oder mehr Zweidrahtleitungen, die zumindest teilweise in Übersprechabstand geführt sind, übertragen werden, **dadurch gekennzeichnet**, daß der Zeitmultiplex-Betrieb (TDM) auf allen Zweidrahtleitungen synchron durchgeführt wird, sodaß auf allen Zweidrahtleitungen gleichzeitig entweder gesendet oder empfangen wird.



1
V.24
2
SCR
3
LCC
TRM
BUFFER
MUX
G.703
SCR
ARQ
BUFFER
CRC GEN.
4
5
ATU-C
6
50
TRANSMITTER FUNCTIONS
7
BYTE/SYMB
ENCODER
8
DMT
MODUL.
9
HP
10
D/A
11
BP
12
13
100
TRANSMITTER
ACTIVATION
G.703
RECEIVE CLOCK
SYSTEM CLOCK
SAMPLE CLOCKS
EXT. TDM SYNC
CLOCK
GENERATION
VCXO
31
BLOCK
SYNC
30
TDM
MANAGER
PILOT
AGC
CLOCK
REC.
20
ATU-R
21
51
PILOT TONE
RECEIVER
ACTIVATION
26
25
ATU-R
24
G.703
DSCR
ARQ BUFFER
CRC VERIFY
ARQ MANAGER
DECODER
SYMB/BYTE
23
EQUALIZER
& UPDATE
22
DMT DE-
MODULAT.
19
HP
17
D/A
16
AGC
15
BP
14
29
V.24
28
DSCR
27
LCC REC
BUFFER
DEMUX
RECEIVER
FUNCTIONS
32
LEVEL
DETECT
18
AGC
CONTROL

FIG.1

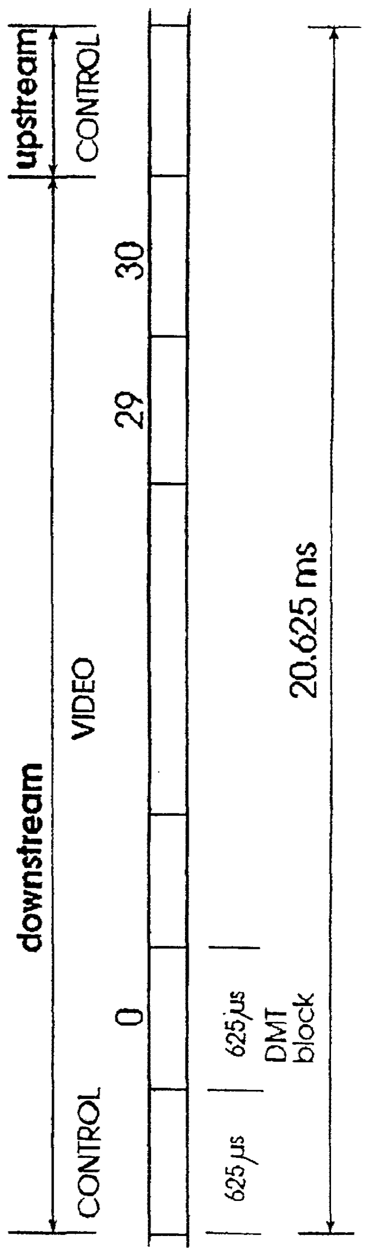

upstream
CONTROL
30
29
downstream
VIDEO
0
CONTROL
625 µs
625 µs
DMT
block
20.625 ms

FIG.2

INTERNATIONAL SEARCH REPORT

Intern   nal Application No
PCT/AT 96/00112

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6   H04L5/14

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6   H04L   H04N

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US,A,4 796 255 (WESTBROOK ET AL.) 3 January 1989 see column 1, line 17 - line 59 see column 2, line 45 - line 68	1,2
A	see figures 1,2 ---	3-7
X	SINGAPORE ICCS '94. CONFERENCE PROCEEDINGS. (CAT. NO.94TH0691-6), PROCEEDINGS OF ICCS '94, SINGAPORE, 14-18 NOV. 1994, ISBN 0-7803-2046-8, 1994, NEW YORK, NY, USA, IEEE, USA, pages 571-575 vol.2, XP002016282 YONG HOON KIM ET AL.: "Dynamic frame control for TDD based wireless LAN" see page 572, paragraph 2.1 see figures 1,2 --- -/--	1,2

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

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"&" document member of the same patent family

1

Date of the actual completion of the international search	Date of mailing of the international search report
18 October 1996	12.11.96

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax (+31-70) 340-3016	Ghigliotti, L

Form PCT/ISA/210 (second sheet) (July 1992)

INTERNATIONAL SEARCH REPORT

Intern. ...onal Application No
PCT/AT 96/00112

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	IEEE TRANSACTIONS ON VEHICULAR TECHNOLOGY, NOV. 1994, USA, vol. 43, no. 4, ISSN 0018-9545, pages 934-945, XP002016283 WAI-CHOONG WONG ET AL.: "Shared time division duplexing: an approach to low-delay high-quality wireless digital speech communications" see paragraph II, pages 935-936 see figure 1 ---	1,2
X	GB,A,2 145 609 (GEN. ELECTRIC CO. PLC) 27 March 1985 see abstract see page 2, line 41 - line 54 see figure 2 ---	1,2
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INTERNATIONAL SEARCH REPORT

Information on patent family members

Intern .ial Application No
PCT/AT 96/00112

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US-A-4796255	03-01-89	NONE		
GB-A-2145609	27-03-85	US-A-	4644525	17-02-87
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C. ALS WESENTLICH ANGESEHENE UNTERLAGEN

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X	US,A,4 796 255 (WESTBROOK ET AL.) 3.Januar 1989 siehe Spalte 1, Zeile 17 - Zeile 59 siehe Spalte 2, Zeile 45 - Zeile 68	1,2
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18.Oktober 1996	12. 11. 96

Name und Postanschrift der Internationale Recherchenbehörde	Bevollmächtigter Bediensteter
Europäisches Patentamt, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax (+31-70) 340-3016	Ghigliotti, L

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C.(Fortsetzung) ALS WESENTLICH ANGESEHENE UNTERLAGEN

Kategorie°	Bezeichnung der Veröffentlichung, soweit erforderlich unter Angabe der in Betracht kommenden Teile	Betr. Anspruch Nr.
X	IEEE TRANSACTIONS ON VEHICULAR TECHNOLOGY, NOV. 1994, USA, Bd. 43, Nr. 4, ISSN 0018-9545, Seiten 934-945, XP002016283 WAI-CHOONG WONG ET AL.: "Shared time division duplexing: an approach to low-delay high-quality wireless digital speech communications" siehe Absatz II, Seiten 935-936 siehe Abbildung 1 ---	1,2
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Angaben zu Veröffentlichungen, die zur selben Patentfamilie gehören

Intern ...ales Aktenzeichen
PCT/AT 96/00112

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INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 : H04M

A2

(11) International Publication Number: WO 99/20027

(43) International Publication Date: 22 April 1999 (22.04.99)

(21) International Application Number: PCT/US98/21442

(22) International Filing Date: 9 October 1998 (09.10.98)

(30) Priority Data:
60/061,689 10 October 1997 (10.10.97) US

(71) Applicant *(for all designated States except US)*: AWARE, INC. [US/US]; 40 Middlesex Turnpike, Bedford, MA 01730 (US).

(72) Inventors; and
(75) Inventors/Applicants *(for US only)*: GROSS, Richard, W. [US/US]; 21 Millett Street, Arlington, MA 02174 (US). GRESZCZUK, John, A. [US/US]; 18 Lowell Drive, Stow, MA 01775 (US). KRINSKY, David, M. [US/US]; 4 Ayer Road, Acton, MA 01720 (US). TZANNES, Marcos [US/US]; 665 Lowell Street, Unit #53, Lexington, MA 02173 (US). TZANNES, Michael, A. [US/US]; 17 Carley Road, Lexington, MA 02173 (US).

(74) Agents: O'DONNELL, Martin, J. et al.; Cesari and McKenna, LLP, 30 Rowes Wharf, Boston, MA 02110 (US).

(81) Designated States: AL, AU, BA, BB, BG, BR, CA, CN, CU, CZ, EE, GE, HU, ID, IL, IS, JP, KP, KR, LC, LK, LR, LT, LV, MG, MK, MN, MX, NO, NZ, PL, RO, SG, SI, SK, SL, TR, TT, UA, US, UZ, VN, YU, ARIPO patent (GH, GM, KE, LS, MW, SD, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published
Without international search report and to be republished upon receipt of that report.

(54) Title: SPLITTERLESS MULTICARRIER MODEM

(57) Abstract

A modem for use in Digital Subscriber Line communications transmits and receives data over the local subscriber loop in common with voice information over the loop, while avoiding the need for voice/data splitters. The modem responds to disruptions associated with "disturbance events" such as on–hook to off–hook transitions and the like by rapidly switching between pre–stored channel parameter control sets defining communications over the loop under varying conditions. In addition to changing parameter control sets responsive to a disturbance event, the modem may also change transmission power levels and other system parameters such as frequency domain equalizer characteristics. Further, provisions are made for reduced bandwidth communications under selected conditions.

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SPLITTERLESS MULTICARRIER MODEM

Cross-reference to related applications:

This application is based in part on the following applications filed by one or more of the inventors herein:

U.S. Provisional Patent Application Serial No. 60/061,689, filed October 10, 1997 by Richard Gross, John Greszcuk, Dave Krinsky, Marcos Tzannes, and Michael Tzannes and entitled "Splitterless Multicarrier Modulation For High Speed Data Transport Over telephone Wires";

U.S. Provisional Patent Application Serial No. **** filed January 16, 1998 by Richard Gross and Michael Tzannes and entitled "Dual Rate Multicarrier Transmission System In A Splitterless Configuration";

U.S. Provisional Patent Application Serial No. *** filed January 21, 1998 by Richard Gross, Marcos Tzannes and Michael Tzannes and entitled "Dual Rate Multicarrier Transmission System In A Splitterless Configuration".

U.S. Provisional Patent Application Serial No. *** filed January 26, 1998 by Richard Gross, Marcos Tzannes and Michael Tzannes and entitled "Multicarrier SystemWith Dynamic Power Levels".

The disclosures of these applications are incorporated by reference herein in their entirety.

Background of the invention

A. Field of the invention.

The invention relates to telephone communication systems and, more particularly, to telephone communication systems which utilize discrete multitone modulation to transmit data over digital subscriber lines.

B. Prior art.

The public switched telephone network (PSTN) provides the most widely available form of electronic communication for most individuals and businesses. Because of its ready availability and the substantial cost of providing alternative facilities, it is increasingly being called upon to accommodate the expanding demands for transmission of substantial amounts of data at high rates. Structured originally to provide voice communication with its consequent narrow bandwidth requirements, the PSTN increasingly relies on digital systems to meet the service demand.

A major limiting factor in the ability to implement high rate digital transmission has been the subscriber loop between the telephone central office (CO) and the premises of the subscriber. This loop most commonly comprises a single pair of twisted wires which are well suited to carrying low-frequency voice communications for which a bandwidth of 0-4 kHz is quite adequate, but which do not readily accommodate broadband communications (i.e., bandwidths on the order of hundreds of kilohertz or more) without adopting new techniques for communication.

One approach to this problem has been the development of discrete multitone digital subscriber line (DMT DSL) technology and its variant, discrete wavelet multitone digital subscriber line (DWMT DSL) technology. These and other forms of discrete multitone digital subscriber line technology (such as ADSL, HDSL, etc.) will commonly be referred to hereinafter generically as "DSL technology" or frequently simply as "DSL". The operation of discrete multitone systems, and their application to DSL technology, is discussed more fully in "Multicarrier Modulation For Data Transmission: An Idea Whose Time Has Come.", IEEE Communications Magazine, May, 1990, pp. 5-14.

In DSL technology, communications over the local subscriber loop between the central office and the subscriber premises is accomplished by modulating the data to be transmitted onto a multiplicity of discrete frequency carriers which are summed together and then transmitted over the subscriber loop. Individually, the carriers form discrete, non-overlapping communication subchannels of limited bandwidth; collectively, they form what is effectively a broadband communications channel. At the receiver end, the carriers are demodulated and the data recovered from them.

The data symbols that are transmitted over each subchannel carry a number of bits that may vary from subchannel to subchannel, dependent on the signal-to-noise ratio (SNR) of the subchannel. The number of bits that can accommodated under specified communication conditions is known as the "bit allocation" of the subchannel, and is calculated for each subchannel in a known manner as a function of the measured SNR of the subchannel and the bit error rate associated with it.

The SNR of the respective subchannels is determined by transmitting a reference signal over the various subchannels and measuring the SNR's of the received signals. The loading information is typically calculated at the receiving or "local" end of the subscriber line (e.g., at the subscriber premises, in the case of transmission from the central telephone office to the subscriber, and at the central office in the case of transmission from the subscriber premises to the central office) and is communicated to the other (transmitting or "remote") end so that each transmitter-receiver pair in communication with each other uses the same information for communication. The bit allocation information is stored at both ends of the communication pair link for use in defining the number of bits to be used on the respective subchannels in transmitting data to a particular receiver. Other subchannel parameters such as subchannel gains, time and frequency domain equalizer coefficients, and other characteristics may also be stored to aid in defining the subchannel.

Information may, of course, be transmitted in either direction over the subscriber line. For many applications, such as the delivery of video, internet services, etc. to a subscriber, the required bandwidth from central office to subscriber is many times that of the required bandwidth from subscriber to central office. One recently developed service providing such a capability is based on discrete multitone asymmetric digital subscriber line (DMT ADSL) technology. In one form of this service, up to two hundred and fifty six subchannels, each of 4312.5 Hz bandwidth, are devoted to downstream (from central office to subscriber premises) communications, while up to thirty two subchannels, each also of 4312.5 Hz bandwidth, provide upstream (from subscriber premises to central office) communications. Communication is by way of "frames" of data and control information. In a presently-used form of ADSL communications, sixty eight data frames and one synchronization frame form a "superframe" that is repeated throughout the transmission. The data frames carry the data that is to be transmitted; the synchronization or "sync" frame provides a known bit sequence that is used to synchronize the transmitting and receiving modems and that also facilitates determination of transmission subchannel characteristics such as signal-to-noise ratio ("SNR"), among others.

Although such systems do in fact provide a significantly increased bandwidth for data communications, special precautions are required to avoid interference with, and from, ordinary voice communications and associated signaling that may be taking place over the subscriber line at the same time that the broadband data is being carried. The signaling activities commonly include, for example, the transmission of ringing signals, busy tone, off-hook indications, on-hook indications, dialing signals, and the like, and the actions commonly accompanying them, e.g., taking the phone off-hook, replacing it on-hook, dialing, etc. These voice communications and their associated signaling, commonly referred to as "plain old telephone service" or POTS , presently are isolated from the data communications by modulating the data communications onto frequencies that are higher than those used for POTS ; the data communications and POTS signals are thereafter separately retrieved by appropriate demodulation and filtering. The filters which separate the data communications and the POTS are commonly referred to as "POTS splitters".

The voice and data communications must be separated at both the central office and the subscriber premises, and thus POTS splitters must be installed at both locations. Installation at the central office is generally not a significant problem, since a single modem at the central office can serve a large number of subscribers, and technicians are commonly available there. Installation at the customer premises is a problem. Typically, a trained technician must visit the premises of every subscriber who wishes to use this technology in order to perform the requisite installation. In connection with this, extensive rewiring may have to be done, dependent on the desired location of the ADSL devices. This is expensive and discourages the use of DSL technology on a widespread basis.

DSL systems also experience disturbances from other data services on adjacent phone lines (such as ADSL, HDSL, ISDN, or T1 service). These services may commence after the subject ADSL service is already initiated and, since DSL for internet access is envisioned as an always-on service, the effect of these disturbances must be ameliorated by the subject ADSL transceiver.

Summary of the invention

A. Objects of the invention

Accordingly, it is an object of the invention to provide an improved digital subscriber line communication system.

Further, it is an object of the invention to provide a digital subscriber line communication system which is compatible with existing voice communication services and which does not require the use of POTS splitters.

Another object of the invention is to provide an improved digital subscriber line communication system that efficiently handles data communications despite random interruptions associated with concurrent carriage of voice communications or disturbances that arise from concurrent data services on adjacent phone lines.

B. Summary description of the invention.

Splitterless Operation

The invention described herein is directed to enhancing the accuracy and reliability of communications in systems using discrete multitone technology (DMT) to communicate data over digital subscriber lines (DSL) in the presence of voice communications and other disturbances. For simplicity of reference, the apparatus and method of the present invention will hereinafter be referred to collectively simply as a modem. One such modem is typically located at a customer premises such as a home or business and is "downstream" from a central office with which it communicates; the other is typically located at the central office and is "upstream" from the customer premises. Consistent with industry practice, the modems are often referred to herein as "ATU-R" ("ADSL Transceiver Unit, Remote", i.e., located at the customer premises) and "ATU-C" ("ADSL Transceiver Unit, Central Office"). Each modem includes a transmitter section for transmitting data and a receiver section for receiving data, and is of the discrete multitone type, i.e., it transmits data over a multiplicity of subchannels of limited bandwidth. Typically, the upstream or ATU-C modem transmits data to the downstream or ATU-R modem over a first set of subchannels, commonly the higher-frequency subchannels, and receives data from the downstream or ATU-R modem over a second, usually smaller, set of subchanels, commonly the lower-frequency subchannels.

Heretofore, such modems have required POTS splitters when used on lines carrying both voice and data. In accordance with the present invention, we provide a data modem for use in discrete multitone communication systems which carry voice and data communications simultaneously and which operate without the special filtering provided by POTS splitters; they are thus "splitterless" modems. In the absence of certain disturbances, referred to herein as "disturbance events" and discussed more fully hereinafter, the modem of our invention transmits data at a rate determined by the transmission capabilities of the system without regard to such disturbances. Preferably, this is the maximum data rate that can be provided for the particular communications subchannel, subject to predefined constraints such as maximum bit error rate, maximum signal power, etc. that may be imposed by other considerations. On the occurrence of a disturbance event on the communications channel, however, the modem of the present invention detects the event and thereupon modifies the subsequent communication operations. Among other responses, the modem changes the bit allocations (and thus possibly the corresponding bit rate) and the subchannel gains among the subchannels, so as to limit interference with and from voice communication activities or to compensate for disturbances from other services or sources sufficiently close to the subject subscriber line as to couple interfering signals into the line. The bit allocations and subchannel gains may be altered for communications in either direction, i.e., upstream, downstream, or both. Effectively, this matches the subchannel capacity to the selected data rate so as to ensure that the pre-specified bit error rate is not exceeded. On cessation of the disturbance event, the system is returned to its initial, high-rate, state.

Disturbance Events

Of particular interest to the present invention are disturbance events that arise from the occurrence of voice communication activities over the data link concurrent with the transmission of data over the link. These activities comprise the voice communications themselves, or activities such as signaling associated with such communications, together with the response to such activities, such as taking a phone off-hook or placing it on-hook. Disturbance events also include other disruptive disturbances such as interference from adjacent phone lines caused, for example, by the presence of other DSL services, ISDN services, T1 services, etc. The cessation of a disturbance event may itself also comprise a disturbance event. For example, the change of a voice communications device such as a telephone from "on-hook" to "off-hook" status can seriously disrupt communications at a modem unless compensated for as described herein or unless otherwise isolated from the modem by means of a POTS splitter as was heretofore done; it is thus a disturbance event that must be dealt with. However, the return of such a device to "on-hook" status can also significantly change the channel characteristics and is therefore also a disturbance event that must be dealt with. The invention described herein efficiently addresses these and other disturbance events.

Channel Control Parameter Sets

In accordance with the present invention, the change in bit allocation is accomplished rapidly and efficiently by switching between stored parameter sets which contain one or more channel control parameters that define data communications by the modem over the subchannels. The parameters sets are preferably determined at the time of initialization of the modem and stored in registers or other memory (e.g., RAM or ROM) in the modem itself, but may instead be stored in devices external to, and in communication with, the modem, e.g., in personal computers, on disk drives etc.

In accordance with one embodiment of this invention, the channel control parameter sets comprise at least a primary set of channel control parameters, stored in a primary channel control table, which defines communications in the absence of voice communication activities or other disturbances; and one or a plurality of secondary sets of channel control parameters, stored in a secondary channel control table, that define data communications responsive to one or more disturbance events. When communicating under control of the primary channel control table, the modem is described hereinafter as being in its "primary" state; when communicating under control of the secondary channel control table, the modem is described hereinafter as being in its "secondary" state. The modem is switched between parameter sets in its primary and secondary states responsive to the occurrence and cessation of disturbance events, as well as among parameter sets in the secondary table responsive to a change from one disturbance event to another. Since the parameter sets are pre-stored and thus need not be exchanged at the time of a disturbance event, the switch is made quickly, limited essentially only by the speed with which the disturbance event is detected and signaled to the other modem participating in the communication, typically not more than a second or so. This greatly reduces the interruption in communications that would otherwise be required by a complete reinitialization of the modems that typically extends over six to ten seconds, and its associated exchange channel control parameters.

As noted previously, in DSL communications, information transmission typically takes place in both directions, i.e. the upstream or ATU-C modem transmits downstream to the ATU-R modem over a first set of subchannels, and the downstream or ATU-R modem transmits upstream to the ATU-C modem over a second, different, set of subchannels. The transmitter and receiver at each modem, accordingly, maintain corresponding channel tables to be used by them in transmitting data to, and receiving data from, the other modem with which it forms a communications pair. Certain parameters such as time and frequency domain equalizer coefficients and echo canceller coefficients are "local" to the receiver with which they are associated, and thus need be maintained only at that receiver. Other parameters such as bit allocations and channel gains are shared with the other modem with which a given modem is in communication (the "modem pair") and thus are stored in both modems, so that during a given communication session, the transmitter of one modem will use the same set of values of a shared parameter as the receiver of the other modem, and vice versa.

In particular, in DSL communications, a key parameter is the number of bits that are to be transmitted over the various subchannels. This is known as the "bit allocation" for the respective subchannels, and is a key element of the primary and secondary parameter sets. It is calculated in a known manner for each subchannel based on the channel SNR, the acceptable bit error rate, and the noise margin of the subchannel. Another important element is the gain for each of the subchannels, and is thus preferably also included in the primary and secondary parameter sets. Thus, each receiver stores a primary channel control table and a secondary channel control table, each of which contains one or more parameter sets that define the subchannel bit allocations to be used by it and by the transmitter of the other modem in communicating with it, and each transmitter also stores a primary channel control table and a secondary channel control table, each of which define the subchannel bit allocations and gains to be used by it for transmission to the other receiver and for reception at that receiver. For the closest match to the actual line over which they are to communicate, those portions of the primary and secondary channel control table at each receiver that define the parameters for use in transmitting to the particular receiver are preferably determined at the modem at which the receiver is located (the "local modem"), as described herein, but it will be understood from the detailed description herein that such tables may also be determined in other ways.

As long as communications over the subscriber line are not impaired by a disturbance event, the modems use the primary channel control table to define communications over the subchannels. When, however, a disturbance event occurs, the modem that detects the event (herein designated "the local modem"; typically, this will be the subscriber modem, ATU-R, particularly in cases of activation of a voice communications device by the subscriber) notifies the other modem of the need to change to the secondary channel control table, and identifies the specific bit allocation set and/or gain set in the secondary table when more than one such set exists. The notification procedure is described in more detail hereinafter. Communications thereafter continue in accordance with the appropriate parameter set (i.e., bit allocations, subchannel gains, and possibly other parameters) from the secondary channel control table. This condition continues until a new disturbance event is detected, at which time the modems revert to the primary channel control table (in the event the disturbance is simply the cessation of communication-disrupting disturbances or interferences) or to a different parameter set secondary channel control table (in the event that the disturbance event is the occurrence of another communication-disrupting disturbance or interference).

In addition to changes in bit allocation among the subchannels, and changes in subchannel gains, further changes may also be made in such communication parameters as time domain equalizer coefficients, frequency domain equalizer coefficients, and the like. These parameters may also be stored in the channel control tables for use in controlling communications, or may be stored in separate tables. Additionally, changes in power level (and corresponding changes in bit allocation and other communication parameters) for communications in either the upstream or the downsteam direction, or both, may be made, and sets of control parameters may be defined on these power levels as well for use in controlling communications. These changes are described in fuller detail below.

As presently contemplated, each modem on the subscribed side of the DSL line will communicate with a corresponding dedicated modem on the central office side. Thus, each central office modem (ATU-C) need store the primary and secondary tables for a specific subscriber only. However, efficiencies may be achieved whenever it is unnecessary to provide service to each subscriber at all times. Under these circumstances, a central office modem may be shared among two or more subscribers, and switched among them as called for. In such a case, the ATU-C will store or have access to a set of channel control tables for each subscriber modem it is to service.

Table Initialization

In the preferred embodiment of the invention, the primary and secondary channel control tables are determined in an initial "training" session ("modem initialization") in which known data is transmitted by one modem, measured on reception by the other, and the tables calculated based on these measurements. Typically, the training session occurs when the modem is first installed at the subscriber premises or at the central office, and the procedure thus "particularizes" the modem to the environment in which it will operate. This environment includes, in addition to the subject data modem, one or more voice communication devices such as telephone handsets, facsimile machines, and other such devices which communicate over a voice frequency subchannel, typically in the range 0-4 kHz. A primary channel control table, comprising a parameter set including at least a set of subchannel bit allocations, and preferably also subchannel gains, is calculated with each device inactive. A secondary channel control table comprising one or more bit communication parameter sets (bit allocations, gains, etc.) is calculated with each voice communication device activated separately, and/or with groups of devices activated concurrently. The tables so determined are then stored at the receiver of one modem and additionally are communicated to the transmitter of the other modem and stored there for use by both modems in subsequent communications.

An alternative approach determines the secondary channel control table (including one or more parameter sets comprising the table) by calculation from the primary channel control table. This is accomplished most simply, for example, by taking one or more of the parameters (e.g., the bit allocation parameter which defines the number of bits to be used for communication across the respective subchannels) as a percentage, fixed or varying across the subchannels, of the corresponding primary parameters; or as determined in accordance with a percentage, fixed or varying across the subchannels, of the SNR's of the respective subchannels; or as determined in accordance with a different bit error rate than provided for in the primary channel control table; or by other techniques.

As a specific example, a number of different sets of bit allocations in the secondary channel control table may be determined as differing percentages (fixed or varying across the subchannels) of the corresponding set of bit allocations in the primary channel control table. Each secondary bit allocation set corresponds to the effect commonly produced by a particular device or class of devices, e.g., a telephone handset, a facsimile machine, etc., as determined by repeated measurements on such devices, and thus may be taken to represent the expected effect of that device over a range of communication conditions, e.g., with a particular type of subscriber line wiring, at a given range from the central office, etc. The subchannel gains may also then be adjusted based on the redetermined bit allocations. The bit allocations and subchannel gains so determined form new secondary parameter sets which may be used responsive to detection of the disturbance events they characterize, and which substitute for determination of the secondary bit allocations and gains on the basis of measurements of the actual disturbances being compensated for.

Alternatively, the secondary channel control table may be determined by adding a power margin to the calculations for each of the entries of the primary table of a magnitude sufficient to accommodate the interference from activation of the voice communications device or from other disturbances. This has the effect of reducing the constellation size for the table entries. The margin may be uniform across the table entries, or may vary across them, as may the percentage factor when that approach is used. Multiple secondary bit allocation sets may be defined by this approach, each based on a different power margin.

One example of the use of varying margins is in response to changes in crosstalk (capacitively coupled noise due to nearby xDSL users, where the "x" indicates the possible varieties of DSL such as ADSL, HDSL, etc.). This crosstalk is, in general, more predictable than signaling events associated with voice communications. The crosstalk spectrum of xDSL sources is well characterized: see, for example, the T1.413 ADSL standard published by the American National Standards Institute. From a primary channel control table associated with a single full initialization, a secondary table comprising a family of bit allocation sets can be calculated, each corresponding to a different crosstalk level. As the number of xDSL systems (and thus crosstalk levels) changes, the ADSL link can quickly switch to one of these automatically generated sets.

The secondary channel control table in the present invention may also be adapted dynamically, e.g., by performing measurements on the transmitted information in each superframe during data communications and monitoring these measurements to determine when the channel performance has sufficiently changed that a different bit allocation set, and possibly different gain set, should be used. We have found that the SNR provides a readily measurable and reliable indicator of the required bit allocations and gains.

In particular, we have found that measurements of the SNR levels across a number of the subchannels during a given communications condition or state provides a "fingerprint" which may reliably be used to quickly select a parameter set, such as the set of bit allocations or the set of gains, for use in subsequent communications during that state. These measurements may be made, for example, on the sync frame that occurs in each superframe or, more generally, during the transmission of reference frames. When the SNR's change by more than a defined amount during communications, the modem at which the measurement is made searches the stored parameter sets for a set whose SNRs on the corresponding subchannels is closest to the measured SNRs, and selects that set for use in subsequent communications. If no parameter set is found within defined limits, the system may be switched to a default state, or a complete reinitialization may be called for, corresponding to a defined pattern of SNR's across some or all of the subchannels, should be used. SNR measurements may also be made on the data carrying signals themselves, i.e., a decision-directed SNR measurement.

Instead of using a multiplicity of secondary subchannel control parameter sets as described above, a simplified approach may construct and use a single secondary set based on a composite of the bit allocation or other characteristics of the individual devices. In one embodiment, the composite is formed by selecting, for each subchannel, the minimum bit allocation exhibited by any device for that subchannel, or the most severe characteristic of any other disturbances, thus forming a single "worst case" set that may be used when any device is activated, regardless of the specific device or disturbance actually present. Or it may be determined as the actual or calculated capacity of the line when all devices are actually or theoretically actuated simultaneously, or all disturbances are present, or both concurrently. Bit allocations sets may also be determined for combinations of subsets of such devices and disturbances. A similar approach may be used to handle the situation where several devices are activated at the same time, and the effects of other disturbances such as cross talk, etc. may also be incorporated into a composite set.

A particular parameter set of the secondary channel control table remains in use for the duration of the session in which the voice device is active or until another change of state occurs, e.g., a further voice device is activated or some other disturbance takes place. When this occurs, the local modem renews its identification procedure to enable determination of the appropriate parameter set for the new conditions. At the end of the session in which the voice device is active, the device returns to inactive (i.e., "on-hook") status and the system reverts to its original ("on-hook") status in which the primary channel control table once again is used for communications between the central office and the subscriber.

Switching the subchannel parameter sets in accordance with the present invention is extremely fast. It can be accomplished in an interval as short as several frames, and thus avoids the lengthy (e.g., several second) delay that would otherwise accompany determination, communication, and switching of newly-determined sets. Further, it avoids communicating new parameter sets at a time when communications have been impaired and error rates are high. Thus, it minimizes disruption to the communication process occasioned by disturbance events.

Detecting Disturbance Events

During subsequent data communications, identification of the device that is activated is achieved in one of a number of ways. In one embodiment of the invention, a specific activation signal is transmitted from the device to the modem on the same side of the subscriber line as the device (referred to herein as "the local modem") on activation of the device. This signal may be transmitted over the communications line to which the device and the local modem are connected or it may be sent over a dedicated connection between the device and the local modem.

In the preferred embodiment of the invention, the local modem monitors the subscriber line to which it and the device are connected and detects a change in line characteristics when the device is activated. For example, the signal to noise ratio (SNR) of the various subchannels can quickly be measured and can be used to identify the particular device that is activated. During multiple sets of initializations, corresponding to multiple communication conditions caused by the devices or by other interferences, the SNR measure for each subchannel is determined for each of the conditions to be tracked (i.e., no devices activated, devices activated separately, two or more devices activated concurrently, adjacent channel interference, etc.) and the measures stored, along with identification of the particular parameter set or sets with which they are associated. When a device is activated, the SNR measurements are used to quickly identify the particular device or devices that have been activated, and the local modem can thereafter switch to the appropriate secondary table.

Disturbance events may also be detected in accordance with the present invention by monitoring selected transmission characteristics that are dependent on these events. These may comprise, in addition to any characteristic SNR accompanying them, such measures as errors in the cyclic redundancy code (CRC) that accompanies transmissions and changes in the error rate of this code; changes in the amplitude, frequency or phase of a pilot tone on the subchannels; or other such indicia. Forward error correction code (FEC) is typically used in ADSL transceivers, and changes in the error rate characteristics of this code, such as how many errors have occurred, how many have been corrected, how many are uncorrected, and the like, can be particularly useful in detecting disturbance events.

In monitoring these characteristics, we distinguish between changes caused by momentary or transient events such as lightning or other such burst noise disturbances, and those associated with disturbance events, the latter continuing for a significant interval (e.g., on the order of seconds or more). In particular, in embodiments that monitor CRC errors or error rates in accordance with the present invention, a switch from one parameter set to another is provided when the errors extend over a number of frames or when the error rate changes by a defined amount for a time greater than a defined minimum. For example, on the occurrence of an off-hook event , a severe form of disturbance to data communications over a subscriber loop, the number of CRC errors suddenly increases and remains at an increased level until it is dealt with. This is distinguished from the occurrence of a transient disturbance such as a lightning strike which causes a momentary increase in CRC errors that does not persist as long as the system has not lost synchronization.

Thus, in accordance with the present invention, the detection of an initial change in the CRC error rate over a number of frames in excess of a defined threshold is one example of the detection of a disturbance event that will result in switching parameter sets. Similar procedures may be undertaken in response to measurement of the signal-to-noise ratio of the subchannel in order to detect a disturbance event based on this characteristic. The decision as to whether a disturbance event has occurred may be based on measurements on a single subchannel; on a multiplicity of subchannels (e.g., the decision to switch parameter sets will be made when more than a defined number of subchannels detect a disturbance event); or the like.

An alternative technique for detecting a disturbance event in accordance with the present invention is the use of a monitor signal, e.g., a pilot tone whose amplitude, frequency, phase or other characteristic is monitored during data transmission. A sudden change in one or more of the monitored characteristics from one frame to another, followed by a smaller or no change in subsequent frames, indicates a disturbance event to which the modem should respond. The monitor signal may comprise a dedicated signal carried by one of the subchannels; a signal carried on a separate control subchannel; a disturbance event itself (e.g., ringing tone, dial tone presence, or other common telephone signals); or other signals.

Communicating The Occurrence of Disturbance Events

After a disturbance event is detected and the appropriate parameter set corresponding to the event is identified, the identification is communicated to the remote modem by means of a selection signal to enable it also to switch to the corresponding parameter set in the secondary table. The selection signal may be in the form of a message transmitted over one or more subchannels or using a predetermined protocol for an embedded operations channel, or it may comprise one or more tones that identify the particular parameter set. ADSL systems use a "guard band" of several subchannels between the sets of subchannels used for upstream and downstream transmission. This guard band may be used to transmit the selection tone or tones. In cases where there is only a single parameter set to be designated, the selection signal may comprise a simple flag (an element that has only two states, i.e., on/off, present/absent, etc.) that is sent to the remote modem to select the set.

In a further embodiment of the invention, use is made of the frame counters at the ATU-R and ATU-C modems that are commonly provided in DSL systems. On detecting a disturbance event, the ATU-R modem notifies the ATU-C modem of the event and specifies a frame at which the change in parameter set, or change in power level and any accompanying change in other parameters, is to take place. The specification may be direct (i.e., the notification specifies a particular frame number at which the change to the secondary table is to be made) or indirect (i.e., on receipt of the notification, the change to the secondary table is made at one of a predetermined number of frames, e.g., the next frame number ending in "0", or in "00", etc., or the nth frame after receiving the notification, where n is some number greater than 0). On reaching the designated frame, both modems (i.e., ATU-R and ATU-C) switch to the new bit allocation set, power level, and other designated parameters.

Alternatively, on detection of a disturbance event, the modems perform a "fast retrain" in order to characterize communications under the new operating conditions and determine a power and/or bit allocation set to be used for the communications. A fast retrain performs only a limited subset of the full initialization procedures, e.g., bit allocation and subchannel gain determination,. The retraining modem (typically the modem local to the disturbance initiating the retraining) then compares the newly determined parameter set with previously stored sets. If the newly-determined set is the same as a previously stored set, a message, flag, or tone is communicated by one modem to the other to designate which of the stored secondary allocation sets is to be used. Otherwise, the newly determined set is used for communications. In the latter event, it must be communicated to the other modem in the communication pair, and communications may be interrupted while this occurs. Nonetheless, on cessation of the event which necessitated a change in parameter sets, the system may simply revert to the primary parameter set, without need for recommunication of that set and thus without further interrupting communications. With proper care in initialization, in most cases a sufficient array of parameter sets may be defined and exchanged at the outset as to avoid the need for subsequent reinitialization in response to most disturbances.

Changing Power Levels

In addition to changing one or more parameter sets in the modem in response to a disturbance event, in accordance with the preferred embodiment of the present invention we also preferably change the communications power level in either the upstream or the downstream direction, or both, in order to further enhance reliable communications. Typically, the change is a reduction in the power level in the upstream direction so as to minimize interference with the voice communications, as well as to reduce echo into the downstream signal, and it will be so described herein. However, it should be understood that there will be some occasions when an increase in power level is called for, such as when interference from adjacent data services requires a higher power level in order to maintain a desired data rate or bit error level, and such a change is accommodated by the present invention in the same manner as that of a decrease. Further, a change in downstream power level may be called for when line conditions change to such an extent that excessive power would otherwise be fed into the downstream channel from the upstream modem

In theory, and in a perfectly linear system, upstream communications activities should have no effect on concurrent voice communications since the two activities occur in separate, non-overlapping frequency bands. However, the telephone system in fact is not a linear system, and nonlinearities in the system can and do inject image signals from the upstream subchannel into the voice subchannel, and possibly into the downstream subchannel as well (i.e., echo), thus producing detectable interference. In accordance with another aspect of the present invention, this effect is reduced below the level of objection by reducing the upstream power level (the power level at which the subscriber or downstream modem transmits to the central office or upstream modem) by a given amount or factor when conditions dictate, e.g., when a voice communications device is off-hook and leakage from the data communications being conducted interferes with the voice communications.

The amount of power reduction may be set in advance. For example, we have found that a nine db reduction in this power (relative to that typically used in ADSL applications using splitters to separate the data and POTS signals) is sufficient in most cases of common interest; under these circumstances, the system operates in one of two alternative power levels at all times. Alternatively, the downstream modem may select one of several different power levels for use, based on the communication conditions prevailing at the time resultant from the disturbance event. For example, the downstream modem may be activated to send a test signal into one or more upstream subchannels and to monitor the leakage (i.e., the echo) of this signal into one or more downstream subchannels as determined, for example, by the SNRs on these subchannels; the power level at which the downstream modem transmits upstream may then be adjusted accordingly in order to minimize the effects of the echo. Commonly, the downstream transmit power is determined by the ATU-R, since the ATU-R is closest to the cause of the disturbance event. In this event, the ATU-R uses a message, flag, or tone to inform the ATU-C of the desired power level to be used for transmission. In either case, at the end of a session, the power level reverts to that used in the "on-hook" state.

In selecting the desired power level, the transmitting modem signals the receiving modem in the communications-pair of the desired change (including the designation of a particular power level from among several power levels, where appropriate), and thereafter implements the change, including switching to a new parameter set associated with that power level. In another embodiment of the invention, the receiving modem detects the power level change at the transmitting modem and switches to a parameter set associated with that power level; upstream communications (i.e., from the ATU-R to the ATU-C) are thereafter conducted at the new power level until the disturbance event (e.g., off-hook condition, etc.) terminates.

While much of the above has been described in terms of a change in power level in the upstream communications from the subscriber modem to the central office modem, it should be noted that a change in power level in the opposite direction may also sometimes be called for. This may be the case, for example, on short subscriber loops (e.g., less than a mile), where the reduced line loss consequent on the greater proximity to the central office may result in the central office initially transmitting at an excessive power level. In such cases, the central office or ATU-C modem performs the role previously performed by the subscriber or ATU-R modem, and vice versa, and a change in power level and other parameters on the downstream communications may be performed as described above. Further, it should also be understood that while it is expected that the power change will most commonly be one that reduces the power level used to communicate, it may in some cases increase it. This will occur, for example, when crosstalk from adjacent services requires an increase in power level of the subject service in order to compensate for the crosstalk.

Changing Other Parameters

A further important change made in response to detecting a disturbance event is a change in the frequency domain equalizers ("FDQ's") associated with each subchannel. These equalizers compensate for the differing distortions (e.g., amplitude loss, phase delay, etc.) suffered by the data during transmission over the subchannel. Typically, they comprise finite impulse response filters with complex coefficients. The coefficients are set during the "initialization" or "training" phase of modem setup. They may subsequently be adjusted based on reference (known) data in reference frames or sync frames transmitted over the communication subchannel. In accordance with the present invention, these filters are adjusted responsive to the transmitted reference data when a disturbance event is detected. The coefficient updating may be performed on all subchannels, or selectively on those whose change in error rates, signal-to-noise ratios, or other error indicia, indicate a disturbance event.

In accordance with one embodiment of the present invention, the coefficients of the frequency domain equalizers for communications both in the absence of a disturbance event or disturbance ("primary FDQ coefficients") and in the presence of such an event or disturbance ("secondary FDQ coefficients") are computed and stored during the initialization or training period. Thereafter, these coefficients are switched responsive to a disturbance event, as is the case with the channel control tables, and are returned to an initial state on the cessation of such an event.

In accordance with another embodiment of the invention, the FDQ coefficients are recomputed responsive to detection of a disturbance event and then used throughout the remainder of the communications session in place of the earlier-stored secondary FDQ tables. The recomputation is accomplished in a short "retrain" session in which known reference data is transmitted between the ATU-R and ATU-C . The received data is compared with the known data and the new FDQ coefficients are determined accordingly. In addition to the frequency domain equalizer coefficients, time domain equalizer coefficients and echo cancellation coefficients may also be determined and stored. Such coefficients are local to the particular receiver, and thus need not be communicated to the other modem of the communications pair. Accordingly, any such retrain will be extremely fast, and any consequent disruption to communication limited.

Excessive Disturbances

In some cases a particular device may cause such interference with communications that compensation for that device by the methods described herein is not practical. This may occur, for example, with antiquated telephones or with particularly complex in-home wiring. In such a case, it is desirable to minimize the disruption caused by such a device by inserting a simple in-line filter between the device and the subscriber line. The filter may comprise, for example, a simple low-pass filter of not more than a cubic inch in volume and a pair of standard connectors such as RJ11 connectors through which the filter connects to the device on one side and to the subscriber line on the other. Unlike POTS splitters, such a connector needs no trained technician to install it, and thus presents no barrier, cost or otherwise, to acceptance of ADSL modems as described herein. Such a device may be detected by measuring the nonlinear distortion of the device when it is activated. This is done by monitoring the echo on the line caused by that device.

Reduced Rate Communications

A further improvement in the operation of the modem of the present invention resides in confining the bandwidth of the downstream transmission to a subset of that normally provided in ADSL communications. This reduces the processing demands on both the local (i.e., central office) and remote (subscriber premises) modems, thereby facilitating the provision of subscriber premises modems at prices more acceptable to consumer, non-business, use; additionally, it further minimizes interference between data transmission and voice communications. For example, limiting the number of subchannels used by the modem to one hundred and twenty eight as opposed to two hundred and fifty six reduces the downstream bandwidth from 1.1 MHz to approximately 552 kHz. When the modem is used with modems that normally provide a greater number of subchannels for such communications, the bit allocations and gains for the subchannels above one hundred and twenty eight are preferably nulled, i.e., set to zero.

The invention is preferably operable with modems that do not have the capabilities described herein, as well, of course, with modems that do. Accordingly, the modem of the present invention identifies its capabilities, preferably during initialization, preparatory to data exchange with another modem. In accordance with the preferred embodiment of the invention, this is preferably done by signaling between the modems that are to participate in communications. The signaling identifies the type of modems in communication and their characteristics of significance to the communication session. For example, one form of ADSL transceiver uses a reduced number of subchannels (typically, thirty two subchannels upstream and one hundred twenty eight subchannels downstream) and provides lower bandwidth communications. A modem having full ADSL capabilities that encounters a reduced-rate modem can then adjust its transmission and reception parameters to match the reduced-rate modem. This may be done, for example, by transmission from one modem to the other of a tone that is reserved for such purposes.

In particular, in accordance with the present invention, on initiation of communications between a central office modem and a subscriber premises modem, the modems identify themselves as "full rate" (i.e., communicating over two hundred and fifty six subchannels) or "reduced rate" (e.g., communicating over some lesser number of subchannels, e.g., one hundred and twenty eight). The communication may be performed via a flag (two-state, e.g., "on/off", "present/absent"), a tone or tones, a message (n-state, n>2), or other form of communication, and may be initiated at either end of the communication subchannel, i.e., either the central office end or the customer premises end.

Brief description of the drawings

The invention description below refers to the accompanying drawings, of which:

Figure 1 is a block and line diagram of a conventional digital subscriber line

(DSL) system using POTS splitters that is characteristic of the prior art;

Figure 2 illustrates an illustrative bit allocation and gains table used in the apparatus of Figure 1;

Figure 3 is a block and line diagram of a splitterless DSL system in accordance with the present invention;

Figure 4 is a block diagram of a splitterless transceiver in accordance with the present invention;

Figures 5A-5C illustrates channel control tables constructed and used in accordance with the present invention;

Figure 6 is a diagram of one form of disturbance event detector in accordance with the present invention;

Figure 7 illustrates the use of a frame counter for communicating the switching decision to the remote modem;

Figure 8 illustrates the preferred procedure used for performing a fast retrain of the modems in accordance with the present invention;

Figure 9 illustrates the manner in which channel control tables may readily be selected in accordance with the present invention; and

Figure 10 illustrates alternative configuration for interconnection of the modems of the present invention.

Detailed description of an illustrative embodiment

Figure 1 shows an ADSL communications system of the type heretofore used incorporating "splitters" to separate voice and data communications transmitted over a telephone line. As there shown, a telephone central office ("CO") 10 is connected to a remote subscriber 12 ("CP: Customer Premises") by a subscriber line or loop 14. Typically, the subscriber line 14 comprises a pair of twisted copper wires; this has been the traditional medium for carrying voice communications between a telephone subscriber or customer and the central office. Designed to carry voice communications in a bandwidth of approximately 4 kHz (kilohertz), its use has been greatly extended by DSL technology.

The central office is, in turn, connected to a digital data network ("DDN") 16 for sending and receiving digital data, as well as to a public switched telephone network ("PSTN") 18 for sending and receiving voice and other low frequency communications. The digital data network is connected to the central office through a digital subscriber line access multiplexer ("DSLAM") 20, while the switched telephone network is connected to the central office through a local switch bank 22. The DSLAM 20 (or its equivalent, such as a data enabled switch line card) connects to a POTS "splitter" 24 through an ADSL transceiver unit -central office ("ATU-C") 26. The local switch 20 also connects to the splitter.

The splitter 24 separates data and voice ("POTS") signals received from the line 14. At the subscriber end of line 14, a splitter 30 performs the same function. In particular, the splitter 30 passes the POTS signals from line 14 to the appropriate devices such as telephone handsets 31, 32, and passes the digital data signals to an ADSL transceiver unit-subscriber ("ATU-R") 34 for application to data utilization devices such as a personal computer ("PC") 36 and the like. The transceiver 34 may advantageously be incorporated as a card in the PC itself; similarly, the transceiver 26 is commonly implemented as a line card in the multiplexer 20.

In this approach, a communication channel of a given bandwidth is divided into a multiplicity of subchannels, each a fraction of the subchannel bandwidth. Data to be transmitted from one transceiver to another is modulated onto each subchannel in accordance with the information-carrying capacity of the particular subchannel. Because of differing signal-to-noise ("SNR") characteristics of the subchannels, the amount of data loaded onto a subchannel may differ from subchannel to subchannel. Accordingly, a "bit allocation table" (shown as table 40 at transceiver 26 and table 42 at transceiver 34) is maintained at each transceiver to define the number of bits that each will transmit on each subchannel to the receiver to which it is connected. These tables are created during an initialization process in which test signals are transmitted by each transceiver to the other and the signals received at the respective transceivers are measured in order to determine the maximum number of bits that can be transmitted from one transceiver to the other on the particular line. The bit allocation table determined by a particular transceiver is then transmitted over the digital subscriber line 14 to the other transceiver for use by the other transceiver in transmitting data to that particular transceiver or to any similar transceiver connected to the line 14. The transmission must, of course, be done at a time when the line is not subject to disturbances which may interfere with communications. This is a significant limitation, and restricts the utilization of this approach.

Referring now to figure 2, a bit allocation table 42 such as is used in the customer premises equipment is shown in further detail. Table 40, used at the central office, is essentially the same in construction and operation and will not further be described. Table 42 has two sections, a first section, 42a, which defines certain communication parameters such as bit allocation capacity and subchannel gain parameters that characterize the respective subchannels and which the transmitter section of transceiver 34 will use in transmitting a signal to the other transceiver (26) with which it is in communication; and a section 42b that defines the parameters that the receiver section of transceiver 34 will use in receiving a signal transmitted from the other transceiver. Communications take place over a plurality of subchannels, here shown, for purposes of illustration only, as subchannels "9", "10", etc. in the transmitter section, and subchannels "40", "41", etc. in the receiver section. In a full-rate ADSL system, there are up to two hundred and fifty six such subchannels, each of bandwidth 4.1 kHz. For example, in one embodiment of the inventin, upstream communications (i.e., from the customer premises to the central telephone office) are conducted on subchannels 8 to 29; downstream communications (from the central office to the customer premises) are conducted on subchannels 32 to 255; subchannels 30 and 31 form a guard band between upstream and downstream communications that may be used for signaling as described hereinafter.

For each subchannel ("SC") 50, a field 52 defines the number of bits ("B") that are to be transmitted over that subchannel by the transmitter of a communications or modem pair, and received by the receiver of that pair, consistent with the prevailing conditions on the subchannel, e.g., measured signal-to-noise ratio (SNR), desired error rate, etc.; column 54 defines the corresponding gains ("G") of the subchannels. A first section, 42a, of the table specifies the bit allocations and gains that transceiver 34 will use in transmitting "upstream" to the transceiver 26; and a second section, 42b, specifies the bit allocations and gains that transceiver 34 will use in receiving transmissions from the transceiver 26. Transceiver 26 has a corresponding table 40 which is the mirror image of table 42, that is, the bit allocations specified for transmission by transceiver 34 are the same as those specified for reception by transceiver 26 and correspondingly for reception by transceiver 34 and transmission by transceiver 26. The table typically may also include a field specifying the gain 54 associated with the particular subchannel.

As noted above, the splitters 24, 30 combine the data and voice communications applied to them for transmission and once again separate these from each other on reception. This is accomplished by means of high pass and low pass filters which separate the low-frequency voice communications from the high-frequency data. The need to utilize such splitters, however, imposes a severe impediment to the widespread adoption of DSL technology by the consumer. In particular, the installation of a splitter at the subscriber premises requires a trip to the premises by a trained technician. This can be quite costly, and will deter many, if not most, consumers from taking advantage of this technology. Nor is incorporating splitters in the communications devices themselves a viable option, since this not only increases the cost of such devices, but requires either the purchase of all new devices or the retrofit of the older devices, which again requires skilled help to accomplish. In accordance with the present invention, we eliminate the splitter at least at the customer premises, thereby enabling adoption and use of DSL modems by the end user without the intervention of trained technical personnel. This, however, requires significant changes in the structure and operation of the DSL transceivers or modems, and the present invention addresses these changes.

In particular, figure 3 shows a DSL transmission system in accordance with the invention in which the composite voice-data signal transmitted from the central office to the subscriber premises is passed to both the subscriber voice equipment 31, 32 and to the data transceiver or modem 34' without the interposition of a splitter at the subscriber premises. In figure 3, components that are unchanged from figure 1 retain the same numbering; components that are modified are designated with a prime superscript. In place of the single table 30 of the transceiver 26 of Figure 1, the transceiver 26' of Figure 3 contains a primary channel control table 41 and a secondary channel control table 43. Similarly, transceiver 34' of Figure 3 contains a primary channel control table 45 and a secondary channel control table 47. It will also be noted that the subscriber side splitter has been eliminated in Figure 3: the reason why this can be done in the present invention will now be described in detail. It will also be noted that the central office splitter 20 in figure 1 has been retained in the configuration of Figure 3: this is optional, not mandatory. Retaining a splitter at the central office can improve the performance somewhat at little cost, since only a single installation is required and that at the central office itself where technical personnel are commonly available in any event. Where this is not the case, it may be eliminated there also.

Turning now to figure 4, the transceiver or modem 34' is shown in greater detail; the modem 26' is essentially the same for present purposes and will not be separately described. As indicated, modem 34' comprises a transmitter module 50; a receiver module 52; a control module 54; a primary channel control table 45; and a secondary channel control table 47. The primary channel control table is shown more fully in figure 5A.; the secondary channel control table is shown more fully in figure 5B.

In figure 5A, the primary channel control table 45 has a transmitter section 45a which stores a primary set of channel control parameters for use in transmitting to a remote receiver over a DSL line; and a receiver section 45b which stores a primary set of channel control parameters for use in receiving communications over a DSL line from a remote transmitter. The subchannels to which the parameters apply are shown in column 45 c. The channel control parameters in the transmitter section 45a include at least a specification of the bit allocations ("B") 45d and preferably also the gains ("G") 45e to be used on the respective subchannels during transmission. The receiver section similarly includes specification of the bit allocations and gains, and preferably also includes specification of the frequency domain equalizer coefficients ("FDQ") 45f, time domain equalizer coefficients ("TDEQ") 45g, and echo canceller coefficients ("FEC") 45h, among others.

Collectively, the parameters: bit allocation, gain, frequency domain coefficient, time domain coefficient, etc. form a parameter set, each of whose members are also sets, e.g. the bit allocation set defining the allocation of bits to each of the subchannels, the gain setting set defining the gains across the subchannels, etc. In accordance with the preferred embodiment of the present invention, the primary channel control table stores a single parameter set which has at least one member, i.e., a bit allocation set, and preferably a gain allocation set as well; this parameter set defines the default communications conditions to which the system will revert in the absence of disturbance events. The secondary channel control table, however, has at least two, and typically more, parameter sets for controlling transmission and reception over the subscriber lines by the respective modems; these sets define communications under various disturbance events which change the default conditions.

In particular, in Figure 5B, the secondary channel control table 47 comprises a plurality of parameter sets 47a, 47b, 47c, etc., of which only three sets are shown for purposes of illustration. Each parameter set includes a transmit portion 47 d and a receive portion 47e. In each portion, one or more parameters are specified, e.g., bit allocations 47f and gains 47g in the transmit portion 47d, and frequency domain coefficients 47h, time domain coefficients 47i, and echo cancellation coefficients 47j in the receive portion 47e. The actual values of the coefficients are typically complex numbers and thus they are represented simply by letters, e.g., "a", "b", etc. in the channel control tables of Figures 5A and 5B. Parameter sets 47b, 47c, and the remaining parameter sets are similarly constructed. As was the case for the primary channel control table, each parameter (e.g., bit allocation) is itself a set of elements that define communication conditions, at least in part, across the subchannels to which they apply and which they help characterize.

The primary channel control table containing a bit allocation parameter set is generated in the usual manner, i.e., during initialization (typically, a period preceding the transmission of "working data" as opposed to test data), known data is transmitted to, and received from, the remote modem with which the instant modem is in communication under the conditions which are to comprise the default condition for the modem. Typically, this will be with all disturbing devices inactivated, so that the highest data rate can be achieved, but the actual conditions will be selected by the user. The data received at each modem is checked against the data known to have been transmitted and the primary channel control parameters such as bit allocation, subchannel gains, and the like are calculated accordingly. This table is thereafter used as long as the system remains undisturbed by disturbance events which disrupt communications over the line.

The secondary channel control table may be determined during initialization in the same manner as the primary table, but with devices that may cause disturbance events actuated in order to redetermine the channel control parameters required for communications under the new conditions. These devices may be actuated one by one, and a secondary parameter control set determined for each and stored in the secondary channel control table; or they may be actuated in groups of two or more, and parameter sets determined accordingly; or various combinations of single and group actuations may be performed and the corresponding parameter sets determined. Secondary parameter sets may similarly be determined from actual measurements with interfering sources such as xDSL transmissions in a common binder with the modems in question, and the resultant sets stored in the secondary table.

Other methods of determination of the secondary table may be employed. For example, one or more secondary parameter sets may be derived from the primary table. Thus, the bit allocation on each subchannel in the secondary table may be taken as a percentage, fixed or varying across the subchannels, of the bit allocation for each subchannel defined in the primary table. Alternatively, it may be calculated from the same data as that of the primary table, but using a larger margin; by using a percentage, fixed or varying across the subchannels, of the signal-to-noise ratio used in calculating the primary table; by providing for a different bit error rate than provided for in the primary; or by other techniques, including those described earlier. Portions of the primary and secondary may be recalculated or improved upon during the communication session, and stored for subsequent use. The calculation or recalculation may be a one-time event or may occur repeatedly, including periodically, throughout a communication session.

Further, although use of a multiplicity of parameter sets in the secondary channel control table will generally provide the best match to the actual channel conditions and thus more nearly approach optimum communications conditions, a simplified second table containing a single composite parameter set may also be used. Thus Figure 5C shows a number of sets 49a-49d of bit allocations for the subchannels 49e and which may represent a corresponding number of different communication devices or conditions associated with communications over these subchannels. A single composite parameter set 49f may be formed as a function of the parameter sets 49a-49d by, for example, selecting, for each subchannel, the minimum bit allocation among the sets 49a-49d for each of the subchannels 49e. Such a set represents a "worst case" condition for activation of any of the devices associated with the sets 49a-49d. Other worst case parameter sets may be formed, for example, on selected groups of devices, thus providing for the case when several devices or disturbances are operating simultaneously.

In the absence of a disturbance event, the transceivers 26', 34' use the primary channel control tables 41, 45 for communications. Responsive to detection of a disturbance event, however, the transceivers 26', 34' switch to one of the parameter sets of the secondary channel control tables 43, 47 to continue the communications under the conditions specified by the particular parameter table. These conditions may specify a diminished bit rate while maintaining the same bit error rate as is provided with the primary channel control table; or may specify the same bit rate but at a higher bit error rate; or may specify a diminished bit rate at a correspondingly diminished power level or margin; or other conditions as determined by the specific channel control tables. On termination of the disturbance condition which caused the switch, the transceivers 26', 34' return to use of the primary tables 41, 45.

Typically, the primary tables provide communications at or near the capacity of the communications channel over line 14. The secondary tables provide communications over the channel at a diminished rate. Switching between the primary and secondary tables (that is, switching from a primary parameter set to a secondary parameter set) in accordance with the present invention is fast: it can be accomplished in an interval as short as several frames (each frame being approximately 250 microseconds in current ADSL systems), and thus avoids the lengthy delay (e.g., on the order of several seconds) that would otherwise be required for determination, communication over the subscriber line, and switching of newly-determined bit allocation tables. Further, it avoids communication of such tables over the subscriber line at a time when communications have been impaired and error rates are therefore high. Thus, utilization of prestored parameter sets in accordance with the present invention minimizes disruption to the communication process occasioned by disturbance events.

The channel control tables are stored in a storage or memory for rapid access and retrieval. Preferably, the storage is a random access memory ("RAM") incorporated into the modem itself, but also comprise such a memory located in other components accessible to the modem, e.g., in a stand-alone memory; in a computer such as a personal computer ("PC"); in a disk drive; or in other elements. Further, the storage may include portions of other forms of memory, such as read only memory ("ROM").

In addition to accessing the channel control tables 45 and 47, the control module 54 of Figure 4 preferably also controls formulation of the secondary control table when this table is calculated on the basis of the primary channel control table. Further, the module 54 monitors the SNR on the subscriber line 14 and calculates the primary and secondary channel control parameter sets when these sets are based on measurement of actual conditions of the line, as will most commonly be the case. To this end, the control module is advantageously implemented as a special purpose digital computer or "DSP" chip particularized to the functions described herein. It may, of course, alternatively be implemented as a general purpose computer or in other fashion, as will be understood by those skilled in the art.

In accordance with the present invention, disturbance events on the subscriber line are distinguished from transient events such as lightning impulses by mean of their consequences. In particular, a signaling event such as an off-hook signal or an on-hook signal typically causes sufficient disruption as to preclude further communications without re-initialization. The event is accompanied by an error code indication that persists throughout the disruption; a change in the amplitude and phase of the physical signal carrying the data or of a pilot tone; the application of a substantial voltage to the line; and other indicia. We monitor the subscriber line for the occurrence of one of more of these characteristics in order to detect the event.

Figure 6 illustrates one manner of detecting a disturbance event in accordance with the present invention. A detector 70, which is preferably included in control module 54, receives signals from line 14 and monitors (step 72) the error code (e.g., CRC errors or the FEC error count) associated with the signals for occurrence of an error indication. If no error is detected (step 74), the detector remains in monitoring mode without further action. If an error is indicated by the error code, a counter is incremented (step 76) and the count is then compared with a predefined threshold (step 78). If the count does not exceed the threshold (step 78, ">N?"), the system remains in monitoring mode and continues to accumulate any detected errors. If the count exceeds the threshold (step 78, Y), the detector emits a "disturbance event" detection signal (step 80) which causes the transceiver in which the detector 70 is located to initiate the process of switching to the appropriate parameter set in the secondary table. The count is reset (line 81) when this occurs.

Instead of monitoring the error code for characteristic behavior (i.e., repeated error over successive frames), in accordance with the present invention one may monitor the amplitude and phase of the physical signals transmitting the data over the subchannel or of a pilot tone transmitted between modems. On the occurrence of a disturbance event, the amplitude and phase of the physical signal undergo significant change, i.e., the amplitude suddenly decreases and the phase suddenly shifts to a new value; thereafter, they maintain approximately their new values during successive frames. This behavior may be monitored as shown in Figure 7 in which a monitor 100 monitors, for example, the amplitude of a data signal or a pilot tone on line 14 and sets a flip-flop 102 to an "active" state ("Q") on detecting a change in the amplitude of greater than a predefined threshold value. Flip-flop 102 enables (input "E") a counter 104 connected to receive counting pulses from a frame counter 106 whenever a new frame is transmitted or received by the modem. These counting pulses are also applied to a threshold counter 108 which accumulates the counts applied to it until it reaches a defined count and then applies the resultant count to a comparator 110 where it is compared with the count in counter 104. If the contents of the counters 104 and 108 are equal, comparator 110 provides an output ("Y") which causes the transceiver to initiate the process of switching to the appropriate table. This also resets the counters 104, 108 and the flip-flop 102. These are also reset (input "R") if the counts of counters 104 and 108 do not match ("N" output of comparator 110).

A similar procedure may be used to generate the table-switching signal based on monitoring the phase change of data signals or pilot tones as noted above. Further, although the operation of the event detector of figure 8 has been explained largely in terms of hardware, it will be understood that it may also readily be implemented in software, or in a combination of hardware and software, as is true of most of the elements described herein.

Still a further approach to detecting a disturbance event is to monitor the disturbance event directly. For example, in the case of off-hook or on-hook signals, a 48 volt dc step voltage is applied to the subscriber line. This signal is sufficiently distinct from other signals as to be readily detectable directly simply by monitoring the line for a step voltage of this size and thereafter generating a table-switching signal in response to its detection. Another approach is to monitor the SNR on one or more subchannels by monitoring the "sync" frames. The presence of a disturbance from data sources on adjacent phone lines manifests itself as a change in the subchannel SNR. A direct method of monitoring disturbance events caused by activation or deactivation of communication-disturbing devices is to directly signal between the device and the local modem on occurrence of either of these events. As shown in Figure 3, for example, signaling lines 35, 37 may be extended directly between the local modem 34' and its associated devices 31, 32 to directly signal a change in these devices, such as their activation ("off hook") or deactivation ("on hook").

In addition to changing the control tables in response to a disturbance event, it is desirable to decrease the upstream transmit power level in order to minimize the interference with the voice communications caused by upstream transmissions, as well as to reduce the leakage of these transmissions into the downstream signal ("echo"). These interferences arise from nonlinearities caused by devices such as telephones that are coupled to the line, especially when the telephones are off-hook. The amount of power reduction required to render the interferences acceptable varies from one telephone to the next. In the preferred embodiment of the invention, a probing signal is used to determine the required decrease in upstream transmit power. In particular, after detecting a disturbance event such as activation or deactivation of a telephone or interference from other sources which can disrupt communications, the transmitter portion of the ATU-R (the "upstream transmitter") transmits a test signal over the subscriber line at varying power levels and measures the echo at the receiver portion of the ATU-R (the "downstream receiver"). The resultant measurement is used to determine an upstream transmission power level that minimizes echo at the downstream receiver or that at least renders it acceptable. The new power level, of course, is typically associated with a corresponding new parameter set in the channel control parameters.

In addition to changing the bit allocation and gain parameters in response to a disturbance event, it is generally necessary to change one or both of the subchannel equalizers, (i.e., the time-domain equalizers or the frequency-domain equalizers), as well as the echo canceller. Appropriate sets of these parameters may be formed in advance in the same manner as the bit allocations and channel gains (i.e., in a preliminary training session, sending test communications over the subscriber line with various devices connected to the line activated, measuring the resultant communication conditions, and determining the various parameters based on the measurements), and stored in the secondary channel control table for recall and use as required. Alternatively, they may be redetermined quickly during a retraining operation following detection of a disturbance event and without excessivley disrupting communications, since these parameters are local to the receiver and thus need not be transmitted to the other modem in the communications pair.

In particular, in accordance with the preferred embodiment of the invention, on detecting a disturbance event, the transceivers enter a "fast retrain" phase, as shown in more detail in Figure 8. A common disturbance event is taking a telephone off hook or replacing it on hook, and this is commonly detected at the ATU-R. The fast retrain process will be illustrated for such an event, although it will be understood that it is not limited to this, and that the retrain may be initiated for any type of disturbance event, and at either end of the communication. Thus, on detecting such an event (Figure 8, event 200), the ATU-R notifies the ATU-C (step 202) to enter the fast retrain mode. The notification is preferably performed by transmitting a specific tone to the ATU-C, but may also comprise a message or other form of communication. On receiving this notification (step 204), the ATU-C awaits notification from the ATU-R of the power levels to be used for subsequent communications. This includes at least the upstream power level, and may include the downstream power level as well, since changing the upstream power level may impact downstream communications to some extent. For purposes of completeness, it will be assumed that both of these power levels are to be changed, although it will be understood that in many cases, only the upstream power level will be changed.

The new power levels to be used are determined by the ATU-R (step 208), which transmits a channel-probing test signal to the upstream transceiver and measures the resultant echo at the downstream receiver; it then sets the upstream power level to minimize the echo into the downstream signal, and may also set the downstream power level to minimize the effects of leakage of the upstream transmission into the downstream transmission at the upstream transmitter. The ATU-R then communicates (steps 210, 212) to the ATU-C the selected upstream and downstream transmission levels, e.g., by transmitting to the upstream transceiver one or more tones modulated by binary PSK (phase shift keying) signals to ensure robust communication of the power levels. The power levels may be specified directly (e.g., as "-30dbm"), or indirectly (e.g., as "level 3" of a predefined group of levels), and the specification may identify the actual value of the power level, or simply the change in power level to be effectuated.

The ATU-R (step 214) and ATU-C (step 216) next commence transmission at the new power levels for purposes of retraining the equalizers and echo cancellers. Preferably, the change to the new power levels is synchronized through use of frame counters which are used in DSL systems to align transmitters and receivers, but the synchronization may be accomplished by other means (e.g., by transmitting a tone or message or by simply sending a flag) or may be left unsynchronized. Based on the training transmission, the ATU-R and ATU-C determine (steps 218, 220) the time and frequency domain equalizer parameters appropriate to the new power levels, as well as the appropriate echo canceller coefficients. The determination may include calculations based on these measurements in order to determine the coefficients, or the measurements may be used to select a particular set or sets of coefficients from one or more precalculated sets stored at the ATU-R and ATU-C, respectively.

For example, as was the case with determination of the power levels responsive to a disturbance event, the SNRs on various subchannels may be used to identify a particular device or devices associated with the event and thus to select an appropriate prestored parameter set stored at the ATU-R and ATU-C, respectively, simply by transmitting to the other modem in the communication pair a message or tone set that specifies the number of the parameter set to be used for subsequent communications. The SNR measurements thus serve as a "signature" of the device or devices associated with the disturbance event, and allow rapid identification of these devices. This approach can significantly reduce the time required to retrain the equalizers and echo cancellers. And even if training is required under particular circumstances, the training time can be meaningfully reduced by using prestored coefficients as the starting point.

To facilitate use of the SNR measurements in retrieving corresponding parameter sets, it is desirable that the various parameter sets as stored be indexed to sets of SNRs, so that one or more parameter sets associated with particular communication conditions may quickly be identified and retrieved. One way in which this may be accomplished is shown in Fig. 9A in which the respective parameter sets such as a first set 250, a second set 252, etc. have, in addition to the subchannel (SC) number 254 and the corresponding bit allocation (BA) and gain (G) entries, a SNR entry 260 characteristic of the parameter set appropriate to a given communication condition, such as "on-hook" (table 250), "off-hook" (table 252), etc. Additional parameter sets such as frequency domain equalizer coefficients, time domain equalizer coefficients, and echo cancellation coefficients may also be stored in the tables, as would be appropriate for the receiver portion of the modem; for the transmitter portion, these coefficients are not applicable and thus are not stored.

An alternative means of linking the subchannel SNRs and the corresponding parameter sets is shown in Figure 9B. As there shown, a simple list structure 270 comprises a parameter set identifier 272, and a multiplicity of SNR measures 274, 276, etc. SNRs for some or all of the subchannels may be included. The list may be searched measure for measure to identify the nearest match to a stored parameter set, and that set then retrieved for subsequent use. In either Figure 9A or 9B the parameter set indexed to the SNRs may be a set of multiple parameters, such as bit allocations and gains, among others, of may comprise a single set such bit allocations only, or gains, only, etc.

The identification of the channel control parameter sets to be used for the subsequent communications is exchanged between the transceivers (steps 226-232) which then switch to these parameter sets (234, 236) and commence communications under the new conditions. The message containing the channel control parameters is preferably modulated in a similar manner as the "power level" message, i.e., using several modulating tones with BPSK signaling. The message is therefore short and very robust. It is important that it be short so that the fast retrain time is minimized, since the modem is not transmitting or receiving data during this time and its temporary unavailability may thus be very noticeable, as would be the case, for example, when the modem is being used for video transmission, or internet access, etc. Similarly, it is important that the message transmission be robust , since error-free communication during a disturbance event is very difficult, due to decreased SNR, impulse noise from ringing or dialing, or the like. Thus, the provision and utilization of pre-stored parameter sets significantly enhances the reliability of communications despite the absence of a splitter at at least one of the modems and despite the presence of disturbance events concurrent with data communications.

It is expected that the modems described herein will most commonly be used in dedicated pairs, i.e., each subscriber (ATU-R) modem will communicate with a dedicated central office (ATU-C) modem. However, in certain cases it may suffice to provide a single master central office modem to service two or more subscriber modems. The present invention accommodates that eventuality as well. Thus, in Figure 10, a central office modem 280 communicates through a switch 282 with a plurality of subscriber modems 284, 286, 288 over subscriber lines 290, 292, 294. The modems may be located at differing distances from the central office and in different communication environments, and thus the channel control tables of each may be unique among themselves. Accordingly, the central office modem stores a master set 296 of individual channel control parameter sets 298, 300, 302, etc., one set (both transmit and receive) for each subscriber modem. On initiating communications to a particular subscriber, the central office modem retrieves the appropriate transmission parameter set for the subscriber and uses it in the subsequent communications. Similarly, on initiating communications to the central office, a given subscriber modem identifies itself to enable the central office modem to retrieve the appropriate reception parameter set for that subscriber.

CONCLUSION

From the foregoing it will be seen that we have provided an improved communications system for communication over subchannels of limited bandwidth such as ordinary residential telephone lines. The system accommodates both voice and data communications over the lines simultaneously, and eliminates the need for the installation and use of "splitters", an expense that might otherwise inhibit the adoption and use of the high communication capacity offered by DSL systems. Thus, it may be implemented and used as widely as conventional modems are today, but offers significantly greater bandwidth than is currently attainable with such modems.

CLAIMS

1. Apparatus for use in connection with a wireline data communication system carrying data in a multiplicity of different frequency bands which may be present concurrently on the line, comprising

A. means for detecting a signaling event associated with at least a first of said bands;

B. means responsive to said detecting means for modifying the processing of signals transmitted over at least a second of said bands.

2. Apparatus for use in connection with a wireline data communication system carrying data in a multiplicity of different frequency bands which may be present concurrently on the line and including means responsive to a signal resulting from a disturbance event to modify the transmission of data over said line.

3. Apparatus according to claim 2 in which said signal is a collection of PSK modulated tones.

4. Apparatus according to claim 2 in which said disturbance event is an on-hook to off-hook transition.

5. Apparatus according to claim 2 in which said disturbance event is off-hook to on-hook transition

6. Apparatus according to claim 2 in which said disturbance event is caused by a change in the crosstalk environment.

7. Apparatus according to claim 2 in which said modification of transmission includes sending a sequence of reference frames.

8. Apparatus according to claim 2 in which said modification of transmissionincludes entering a fast retrain mode.

9. Apparatus according to claim 1 in which said detecting means comprises

(1) means for measuring, at a multiplicity of different times, a characteristic of a signal transmitted over said wireline

(2) means for activating said modifying means when samples of the measured characteristics differ in a defined manner at selected different times.

10. Apparatus according to claim 9 in which said measuring means measures the extent of errors in error-correcting code associated with the signals whose processing is to be modified and activates said modifying means only when the extent of said errors exceeds a defined threshold for at least a defined number of times.

11. Apparatus according to claim 10 in which said measuring means activates said modifying means only when the number of errors in each said sample exceeds a defined number in each of two or more samples.

12. Apparatus according to claim 9 in which said measuring means measures a characteristic of signals transmitted over a plurality of different frequency bands and activates said modifying means only when the measured characteristic exceeds defined thresholds associated with each of said plurality of frequency bands.

13. Apparatus according to claim 1 in which said wireline data communication system comprises a telephone subscriber loop carrying both voice and data signals, and in which said signaling event comprises an off-hook event.

14. Apparatus according to claim 1 in which said wireline data communication system comprises a telephone subscriber loop carrying both voice and data signals, and in which said signaling event comprises an on-hook event.

15. Apparatus according to claim 14 which includes a frequency domain equalizer for equalizing the frequency characteristics of each of said frequency bands in accordance with reference signals transmitted over said bands and in which said modifying means comprises means for changing the characteristics of said equalizers in accordance with

measurements on said reference signals.

16. Apparatus according to claim 9 in which said measuring means measures the signal-to-noise ratio of said reference signals and activates said modifying means only when said ratio is less than a defined threshold for at least a defined number of times

17. Apparatus according to claim 9 in which said data communication system includes means for transmitting a pilot tone and in which said apparatus includes means for measuring at least one characteristic of said tone at different times and means for activating said modifying means only when said characteristics manifest changes exceeding a defined threshold for at least a defined number of times.

18. Apparatus according to claim 9 which includes means for transmitting over said wireline information back to a source of said information signals, said means transmitting at a first power level in the absence of detection of a signaling event, and transmitting at a different power level responsive to detection of a signaling event.

19. Apparatus according to claim 9 which includes a first set of stored parameters for use in processing said information when said system is in a first state.

20. Apparatus according to claim 19 which further includes a second set of stored parameters for processing said information when said system switches to a second state responsive to detecting a signaling event.

21. Apparatus according to claim 20 in which said second set is precomputed.

22. Apparatus according to claim 21 in which said second set is computed responsive to reference signals received on said subchannel subsequent to detection of a signaling event.

23. Apparatus according to claim 21 in which said first and second sets are computed on initiating a communications session.

24. Apparatus according to claim 1 including means for varying the data rate at which said modifying meansprocesses said signals.

25. In a modem communicating data over a multiplicity of discrete sub-subchannels, each

characterized by a bit allocation parameter defining the allocation of bits to the corresponding subchannel for communication over said subchannel, the improvement comprising:

A. means for storing a first channel control table for allocating bits to said subchannel during a first communication condition;

B. means defining a second channel control table for allocating bits to said table during a second communication condition;

26. A modem according to claim 25 which includes a

means for switching between said tables on the detection of a defined event.

27. A modem according to claim 25 in which said first table establishes the communications capabilities of said modem during normal operation.

28. A modem according to claim 27 in which said second table establishes the communications capabilities of said modem during diminished operation.

29. A modem according to claim 25 in which said defined event includes signaling events comprising transitions between on-hook and off-hook conditions.

30. A modem according to claim 29 in which said first table defines communications in the absence of a signaling event.

31. A modem according to claim 30 in which said second table defines communications responsive to detection of a signaling event.

32. A modem according to claim 31 in which said switching means switches from said second table to said first table on detection of a signaling event indicative of cessation of a previously-detected signaling event.

33. A modem according to claim 25 in which said first and second tables are determined during an initialization session in which the communication capabilities of said sub-subchannels are determined.

34. A modem according to claim 33 in which said first table is determined in the absence of interfering signaling conditions.

35. A modem according to claim 34 in which said second table is determined as a function of said first table.

36. A modem according to claim 35 in which the bit allocations of said second table are determined as a percentage of the bit allocations of said first table.

37. A modem according to claim 27 in which the bit allocations of second table are determined by adding noise margins to the determination of the bit allocations of the corresponding sub-subchannels of said first table.

38. A modem according to claim 25 in which said second channel control table is determined responsive to a plurality of signaling events created by a corresponding plurality of event-generating sources, each defining a channel control table specific to the given source, and comprises a composite table formed by selecting, for each sub-subchannel, the minimum bit allocation for the corresponding sub-subchannel of the table associated with each of the plurality of sources.

39. A modem according to claim 25 in which said second channel control table is selected from a plurality of tables determined responsive to a plurality of signaling events created by a corresponding plurality of event-generating sources, each defining a channel control table specific to the given source.

40. A modem according to claim 39 which includes means for selecting one of said plurality of tables for use as said second table in accordance with the source generating an event.

41. A modem according to claim 25 which further includes:

C. means for redetermining said channel control tables while said modem is in either of said communication conditions; and

D. means for communicating a redetermined table to a second modem engaged in communication with said modem.

42. A modem according to claim 41 in which said communicating means communicates said redetermined table over a dedicated sub-subchannel selected from among said discrete sub-subchannels.

43. A modem according to claim 41 in which said communicating means further communicates to said second modem information identifying the type of said redetermined table.

44. A modem for use in asymmetric digital subscriber loop communications having both upstream and downstream communication subchannels formed from a plurality of sub-subchannels, said loop adapted to carry both voice and data communications thereon, comprising:

A. means for storing a first table defining data communications between said modem and a second modem connected to said loop during a first communication state;

B. means for storing a second table defining data communications between said modem and said second modem during a second communication state.

45. A modem according to claim 44 that includesmeans for switching between said tables responsive to the occurrence of selected events.

46. A modem for use in asymmetric digital subscriber loop communications having both upstream and downstream communication subchannels formed from a plurality of sub-subchannels, said loop adapted to carry both voice and data communications thereon, comprising:

A. means for storing a first table defining data communications between said modem and a second modem connected to said loop during a first communication state;

B. means for storing a second table defining data communications between said modem and said second modem during a second communication state; and

C. means for selecting between said tables based on signals received from said second modem.

47. A modem according to claim 44 which includes:

D. means for detecting said selected events, said means including

(1) means for monitoring a selected characteristic of at least one of said communication subchannels during a plurality of communication intervals;

(2) means for determining differences in the selected characteristic over said plurality of intervals;

(3) means for generating a signal initiating switching of said tables when said differences exhibit a defined pattern.

48. A modem according to claim 47 in which said pattern comprises an initial difference above a first threshold amount followed by at least a subsequent differences less than a second threshold amount.

49. A modem according to claim 48 in which said first threshold is greater than said second threshold.

50. A modem according to claim 49 in which said pattern comprises an initial difference above a first threshold amount followed by a plurality of subsequent differences less than a second threshold amount.

51. A modem according to claim 48 in which said selected characteristic is monitored over at least one sub-subchannel.

52. A modem according to claim 48 in which said selected characteristic is monitored over a plurality of sub-subchannels.

53. A modem according to claim 52 which includes means for averaging the monitored values of said selected characteristic over said sub-subchannels for use in comparing said initial difference to said first threshold.

54. A modem according to claim 52 which includes means for averaging the monitored

values of said selected characteristic over said sub-subchannels for use in comparing said subsequent difference to said second threshold.

55. A modem according to claim 49 in which said characteristic comprises an error code error.

56. A modem according to claim 49 in which said characteristic comprises a signal-to-noise ratio.

57. A modem according to claim in which said characteristic comprises a parameter of a pilot tone.

58. A modem according to claim 44 in which said first table establishes a data rate greater than that of said second table.

59. A modem according to claim 58 in which said tables define the number of bits transmitted over the respective sub-subchannels.

60. A modem according to claim 59 in which said events comprise signaling events selected from the group comprising off-hook, on-hook, ringing, and busy.

61. A modem according to claim 47 in which said switching means returns said modem to said first communication state on termination of the event causing the switching.

62. A modem according to claim 44 which includes:

D. means for emitting into said loop a test signal for probing the return characteristics of transmissions into the loop by said modem; and

E. means for limiting the power level of said transmissions in accordance with the measured return characteristics.

63. A modem according to claim 62 in which said probe comprises a tone at a defined amplitude and frequency and in which the measured return characteristics comprise at least one characteristic selected from the group comprising the amplitude and frequency of the signal returned to said modem in response to emission of said tone.

64. A modem according to claim 62 in which said probe comprises a plurality of tones at defined amplitudes and frequencies and in which the measured return characteristics comprise at least one characteristic selected from the group comprising the amplitudes and frequencies of the signal returned to said modem in response to emission of said tone.

65. A modem according to claim 44 which includes equalizers for equalizing the transmission characteristics of said subchannels and in which said tables define:;

(1) coefficients of time domain equalizers or

(2) coefficients of frequency domain equalizers or

(3) coefficients of digital echo cancellers

66. A modem according to claim 44 in which said first table is determined during an initialization process in the absence of a selected event.

67. A modem according to claim 66 in which said second table is determined during an initialization process in the presence of a selected event.

68. A modem according to claim 67 in which said second table is redetermined responsive to occurrence of a selected event.

69. A modem according to claim 68 in which redetermined tables are communicated from a given modem to other modems with which it is in communication during a quiescent state.

70. A modem according to claim 47 in which said generating means causes transmission of a switch-control signal over one of said sub-subchannels in response to detection of a selected event.

71. A modem according to claim 47 in which said generating means causes transmission of a tone in response to detection of a selected event.

72. Apparatus for use in communicating digital data over a digital subscriber line concurrent with voice communications over said line, comprising:

A. a transceiver for communicating digital data to and from said line;

B. a first storage element for storing a first set of communication parameters for use in communicating data under a first communication condition; and

C. a second storage element for storing a second set of communication parameters for use in communicating data under a second communication condition.

73. Apparatus according to claim 72 including a means for monitoring communication conditions on said line and for switching between said first and second sets of communication parameters responsive to changes between said communication conditions.

74. Apparatus according to claim 72 including means responsive to signals communicated to it to switch between said sets of communicaton parameters.

75. Apparatus according to claim 72 which communicates said data over a plurality of subchannels of different frequency and at least potentially different information-carrying capacity and in which said communication parameters comprise at least a channel control table defining the number of bits to be allocated to the subchannels for communications under the respective conditions.

76. Apparatus according to claim 72 which communicates said data over a plurality of subchannels of different frequency and at least potentially different information-carrying capacity and in which said communication parameters comprise subchannel gain tables defining the gain characteristics of the subchannels for communications under the respective conditions.

77. Apparatus according to claim 72 which communicates said data over a plurality of subchannels of different frequency and at least potentially different information-carrying capacity and in which said communication parameters comprise frequency domain equalizers defining the frequency characteristics of the subchannels for communications under the respective conditions.

78. Apparatus according to claim 72 in which both sets of communication parameters are determined during an initialization interval preceding communication of working data.

79. Apparatus according to claim 72 in which said first set of communication parameters is determined during an initialization interval preceding communication of working data and said second set of parameters is determined during a subsequent interval following communication of working data and characterized by said second communications conditions.

80. Apparatus according to claim 79 in which said second set of communication parameters is determined at a first transceiver of a transceiver pair in communication with each other and is communicated to a second transceiver in said pair during a time when said transceivers are operating with an earlier set of set of secondary parameters.

81. Apparatus according to claim 80 in which said transceivers revert to said first set of communications parameters responsive to return of communications to a first communications condition.

82. Apparatus according to claim 72 which includes means for signaling between said transceivers a desired change in communications parameters.

83. Apparatus according to claim 82 in which said signaling means comprises means for transmitting messages over one or more subchannels.

84. Apparatus according to claim 82 in which said signaling means comprises means for transmitting messages over one or more subchannels intermediate subchannels used for upstream and downstream communications.

85. Apparatus according to claim 83 in which said messages comprise tones.

86. Apparatus according to claim 72 in which said transceiver transmits and receives data over a defined number of subchannels and which includes means for identifying the subchannels over which said transceivers will communicate with each other.

87. Apparatus according to claim 86 in which said identifying means includes means for nulling at least those portions of the stored sets of communications parameters that define the bit capacity of the subchanels that are being excluded from communications.

88. Apparatus according to claim 72 in which said second set of parameters includes communication parameters corresponding to a plurality of devices connected for voice communications over said line.

89. Apparatus according to claim 80 in which said second set of parameters includes a plurality of subsets of communications parameters characteristic of a corresponding plurality of voice communication devices for defining communications when a selected device is active.

90. Apparatus according to claim 89 including means for identifying which of said plurality of devices is active and for selecting the corresponding communications parameter set for such device.

91. Apparatus according to claim 90 in which said identifying means includes signaling means interconnecting said voice communication devices to said transceiver.

92. In a communication system using discrete multitone modulation, the improvement comprising storing a first channel control table for use in defining communications under a first communication state and storing at least a second channel control table for communication under a second communication state.

93. A modem for use in symmetric or asymmetric digital subscriber loop communications having both upstream and downstream communication subchannels formed from a plurality of sub-subchannels, comprising:

A. means for storing a first table defining data communications between said modem and a second modem connected to said loop during a first communication state;

B. means for storing a second table defining data communications between said modem and said second modem during a second communication state:

94. A modem according to claim 93 that includes means for switching between said tables responsive to the occurrence of selected events.

95. A modem according to claim 94 in which said selected event includes a transition

from on-hook to off-hook.

96. A modem according to claim 94 in which said selected event includes a transition from off-hook to on-hook.

97. A modem according to claim 94 in which said selected event includes a change in the crosstalk environment.

98. A modem according to claim 93 that includes means for switching between said tables based upon reception of a signal from a remote modem.

99. A modem according to claim 98 in which said signal includes a message.

100. A modem according to claim 98 in which said signal includes a tone or set of tones.

101. A modem according to claim 98 in which said signal includes a flag.

102. A modem according to claim 93 that includes means for switching between said tables at a time that depends upon a frame counter.

103. A modem according to claim 93 that includes means for switching between said tables at a time that depends upon a flag.

104. A multicarrier modem for use in symmetric or asymmetric digital subscriber loop communications having both upstream and downstream communication subchannels formed from a plurality of subchannels that includes a means to select the number of said subchannels that are to be used for communications based upon a signal from a remote modem.

105. A modem according to claim 104 in which said signal is received prior to initialization of modem.

106. A modem according to claim 104 in which said signal is a message dictating how many subchannels are to be used.

107. A modem according to claim 104 in which said signal is a message selecting one of a collection of candidate subchannel selections.

108. A modem according to claim 104 in which said signal is a tone or collection of tones selecting one of a collection of candidate subchannel selections.

109. A multicarrier modem for use in symmetric or asymmetric digital subscriber loop communications having both upstream and downstream communication subchannels formed from a plurality of subchannels that includes a means to signal to a remote modem the number of said subchannels that are to be used for communications.

110. A modem according to claim 109 in which said signal is transmitted prior to initialization of modem.

111. A modem according to claim 109 in which said signal is a message dictating how many subchannels are to be used.

112. A modem according to claim 109 in which said signal is a message selecting one of a collection of candidate subchannel selections.

113. A modem according to claim 109 in which said signal is a tone or collection of tones selecting one of a collection of candidate subchannel selections.

114. A multicarrier modem for use in symmetric or asymmetric digital subscriber loop communications having both upstream and downstream communication subchannels formed from a plurality of subchannels, comprising of a means to limit the number of transmission subchannels in order to communicate with a remote modem that is only capable of receiving the limited frequency band.

115. A multicarrier modem for use in symmetric or asymmetric digital subscriber loop communications having both upstream and downstream communication subchannels formed from a plurality of subchannels, comprising of a means to limit the number of receiver subchannels in order to communicate with a remote modem that is only capable of transmitting the limited frequency band.

116. A multicarrier modem that for use in symmetric or asymmetric digital subscriber loop communications having both upstream and downstream communication subchannels

formed from a plurality of subchannels, comprising of a means to determine the location of a telephone that would benefit from the use of a low pass filter.

117. A multicarrier modem according to claim 116 in which said determination means includes monitoring the signal to noise ratio when said telephone goes off-hook.

118. A multicarrier modem according to claim 116 in which said determination means includes monitoring the echo response of the transmitted signal when said telephone goes off-hook.

119. In a modem communicating data over a wireline via a multiplicity of discrete sub-channels in accordance with a bit-loading specification defining the allocation of bits to the corresponding subchannel for communication thereon, the improvement comprising:

A irst means for storing a primary bit allocation table for allocating said bits during a first communication condition; and

B. second means for storing a secondary bit allocation table for allocating said bits during a second communication condition.

120. A modem according to claim 119 which includes means for switching between bit allocation sets defined by said tables.

121. A modem according to claim 120 in which said switching means is actuated respon-sive to at least one of the events comprising receipt of a message, a tone, or a flag from a remote modem.

122. A modem according to claim 121 in which switching means includes the use of a frame counter to designate when said switch is to occur.

123. A modem according to claim 119 in which said primary bit allocation table defines communications in the absence of a disturbance event, and in which said secondary bit allocation table defines communications in response to said disturbance event.

124. A modem according to claim 123 in which said secondary bit allocation table defines communications over said subchannels for times when said subchannels are affected by

voice communication activities.

125. A modem according to claim 124 in which said secondary bit allocation table defines communications over said subchannels for times when said subchannels are affected by voice communication devices that have entered the off-hook state.

126. A modem according to claim 124 in which said primary bit allocation table defines communications over said subchannels for times when said subchannels are affected by voice communication devices that have returned from an off-hook state.

127. A modem according to claim 119 in which said primary table is determined in a preliminary training session in which potentially interfering voice communication devices connected to the line are inactive.

128. A modem according to claim 119 in which said primary table is determined in the absence of disturbance events.

129. A modem according to claim 119 in which said primary bit allocation table is determined in advance of installation of said modem.

130. A modem according to claim 119 in which said secondary table is determined in an initial training session based on measurements of communications over said wireline.

131. A modem according to claim 119 in which said secondary table is determined in initial training sessions based on measurements of communications over said wireline with potentially interfering voice communication devices connected to the line selectively activated to thereby form a secondary table comprising a plurality of bit allocation sets corresponding to the plurality of activated devices.

132. A modem according to claim 131 in which said devices are activated one by one so that each bit allocation set corresponds to a single device.

133. A modem according to claim 131 in which said devices are activated in groups of two or more so that each bit allocation set corresponds to one of said groups.

134. A modem according to claim 119 in which said secondary bit allocation table is de-

termined from said primary bit allocation table.

135. A modem according to claim 119 in which the bit allocations of said secondary table are determined as a percentage of the bit allocations of said primary table.

136. A modem according to claim 119 in which the bit allocations of said secondary table are determined based on a percentage of the signal to noise ratios on which the bit allocations of said primary table are determined.

137. A modem according to claim 119 in which the bit allocations of said secondary table are determined based on information defining said primary table but using a different bit error rate

138. A modem according to claim 119 in which said secondary bit allocation table is formed as a composite of the bit loading sets of a multiplicity of voice communication devices and/or disturbances.

139 A modem according to claim 119 in which the bit allocation value for each subchannel in said composite is the worst-case value for the corresponding subchannel in the bit allocation sets defining said devices and/or disturbances.

140. A modem according to claim 119 in which said secondary bit allocation table is determined by adding a power margin to the calculations for the respective entries of the primary table.

141. A modem according to claim 119 in which said secondary table comprises a set of bit allocation tables defining the bit allocations for a corresponding set of devices that may be connected to said wireline.

142. A modem according to claim 119 in which said secondary table comprises a set of bit allocation tables defining the bit allocations for a corresponding set of disturbances on said wireline.

143. A modem according to claim 119 in which said secondary table comprises a set of bit allocation tables defining the bit allocations for a corresponding set of devices and

disturbances on said wireline.

144. A modem according to claim 119 in which a plurality of secondary bit allocation tables are determined by adding a corresponding plurality of power margins to the calculations for the respective entries of the primary table, each secondary table so determined corresponding to a different communications state.

145. A modem according to claim 119 in which said power margin is substantially uniform across the entries of a table.

146. A modem according to claim 119 in which said power margin varies across the entries of a table.

147. A modem according to claim 119 configured to switch to a secondary state corresponding to use of said secondary bit allocation table for communications responsive to occurrence of a disturbance event.

148. A modem according to claim 147 configured to switch to a primary state corresponding to use of said primary bit allocation table for communications responsive to cessation of a disturbance event.

149. A modem according to claim 147 configured to switch to a different secondary state corresponding to use of a different set of bit allocations in said secondary bit allocation table for communications responsive to occurrence of a further disturbance event, different from a disturbance event preceding it, while said modem is in said secondary state.

150. A modem according to claim 119 in which said switching means includes means responsive to a disturbance event to thereby initiate a switch between said tables.

151. A modem according to claim 150 which includes a signaling line connecting a device to said modem for signaling to said modem the occurrence of a disturbance event.

152. A modem according to claim 150 which includes means for detecting a disturbance event on said line.

153. A modem according to claim 152 in which said detecting means includes means for

monitoring the signal to noise ratios on one or more subchannels of said line and means responsive to said ratios for selecting a bit allocation set for use in communications.

154. A modem according to claim 152 in which said detecting means includes means for monitoring a parameter of a tone or collection of tones and means responsive to said parameter for selecting a bit allocation set for use in communications.

155. A modem according to claim 152 in which said parameter includes the amplitude and/or phase of said tone or tones.

156. In a modem communicating data over a wireline via a multiplicity of discrete subchannels in accordance with a gain specification defining the allocation of gains to the corresponding subchannel for communication thereon, the improvement comprising:

A. first means for storing a primary gain set for allocating said gains during a first communication condition; and

B. second means for storing a secondary gain set for allocating said gains during a second communication condition.

157. A modem according to claim 156 which includes means for switching between said gain sets.

158. A modem according to claim 157 in which said switching means is actuated responsive to at least one of the events comprising receipt of a message, a tone, or a flag from a remote modem.

159. A modem according to claim 157 in which said switching means is actuated responsive to its detection of a disturbance event.

160. A discrete multitone modem including a transmitter for communicating to a remote receiver at one of a plurality of power levels associated with particular communication conditions on a digital subscriber line, comprising

A. means for monitoring at least one parameter indicative of communication conditions on said line;

B. means dependent on said parameter for selecting the power level at which said modem either transmits, or receives, data or both.

161. A discrete multitone modem including a transmitter for communicating to a remote receiver at one of a plurality of power levels associated with particular communication conditions on a digital subscriber line and adapted to receive a power select signal indicating a power level to be used for subsequent transmissions.

162. A discrete multitone modem according to claim 160 which includes means for communicating to another modem with which it communicates a power select signal indicating a power level to be used for subsequent transmissions.

163 A discrete multitone modem according to claim 160 which includes means for receiving from another modem with which it communicates a power select signal indicating a power level to be used for subsequent transmissions.

164 A discrete multitone modem according to claim 160 in which said power select signal identifies a specific power level at which said other modem is to receive data from it.

165. A discrete multitone modem according to claim 162 in which said power select signal identifies a specific power level at which said other modem is to transmit data to it.

166. A discrete multitone modem according to either of claims 164 or 165in which said discrete power level comprises one of several predefined power levels for communication between said modems.

167. A discrete multitone modem according to claim 162 in which the means for communicating said power select signal includes means for transmitting said signal over at least one subchannel intermediate an upstream and a downstream set of data subchannels over which said modem communicates.

168. A discrete multitone modem according to claim 167 which the means for communicating said power select signal includes means for transmitting said signal over one or more data subchannels over which said modem communicates.

169. A discrete multitone modem according to claim 160 which includes a plurality of parameter sets stored in said modem and defining communications under a plurality of different communication conditions on said line.

170. A discrete multitone modem according to claim 169 in which said parameter sets include at least a primary set of parameters for controlling communications in the absence of a disturbance event, and a secondary set for controlling communications responsive to a disturbance event.

171. A discrete multitone modem according to claim 169 in which said monitoring means monitors the signal to noise ratio on one or more subchannels over which said modem communicates and selects a parameter set based on said ratio for controlling subsequent communications.

172. A discrete multitone modem according to claim 169 in which said parameter sets include a set of parameters defining the power level at which said modem transmits to other modems.

173. A discrete multitone modem according to 169 in which said parameter sets include a set of parameters defining the power level at which said modem receives communications from other modems.

174. A discrete multitone modem according to claim 173 in which said modem includes means for transmitting to another modem with which it is in communication a signal indicating a parameter set to be used in subsequent communications between said modems.

175. A discrete multitone modem according to claim 172 in which said modem includes means for receiving from another modem with which it is in communication a signal indicating a parameter set to be used in subsequent communications between said modems.

176. A discrete multitone modem according to claim 160 in which said modem communicates to another modem a desired power level by itself changing the power level at which it communicates with said other modem.

177. A discrete multitone modem including a transmitter for communicating to a remote receiver at one of a plurality of power levels associated with particular communication conditions on a digital subscriber line and storing a plurality of sets of channel control parameters corresponding to said power levels, comprising

A. means responsive to a disturbance event to select a power level at which said transmitter transmits to said receiver; and

B. means for communicating the selected power level to said receiver.

178. A discrete multitone modem including a transmitter for communicating to a remote receiver at one of a plurality of power levels associated with particular communication conditions on a digital subscriber line and storing a plurality of sets of channel control parameters corresponding to said power levels and adapted to receive a power select signal indicating a power level to be used for subsequent transmissions.

179. A discrete multitone modem according to claim 177 in which the means for communicating the change in power level transmits a power power select signal to the remote receiver indicative of the change in power level.

180. A discrete multitone modem according to claim 179 in which the transmitting means transmits a tone indicating the desired change in power level to the remote receiver.

181. A discrete multitone modem according to claim 179 in which the transmitting means transmits a plurality of tones indicating the desired change in power level to the remote receiver.

182. A discrete multitone modem according to claim 181 in which the plurality of tones designates a particular one of several power levels to which the remote receiver is to switch.

183. A discrete multitone modem according to claim 179 in which the means for communicating the change in power level designates a particular one of several power levels to which the remote receiver is to switch.

184. A discrete multitone modem according to claim 179 in which the means for communicating the change in power level to the remote receiver includes means for transmitting a power power select signal over at least one subchannel intermediate an upstream and a downstream set of data subchannels over which said modem communicates.

185. A discrete multitone modem according to claim 177 in which the means for communicating the change in power level to the remote receiver comprises

(1) means associated with the transmitter for effectuating the change in power level at said transmitter;

(2) means in the remote receiver responsive to the change in power level at the transmitter for changing the power level of its reception in accordance therewith.

186. A discrete multitone modem according to claim 177 in which the means for communicating the change in power level to the remote receiver transmits to the remote receiver a frame count at which the remote receiver is to effectuate the change in power level.

187. A discrete multitone modem according to claim 178 in which the means for receiving the power select signal inlcudes a frame count at which said modem is to effectuate the change in power level.

188. A discrete multitone modem according to claim 177 including a receiver responsive to communication of a power level change from a remote transmitter to thereby:

(1) measure at least one parameter indicative of communication conditions on said line responsive to said power level change, and

(2) select new channel control parameters from a plurality of sets of prestored channel control parameters based on said measurement.

189. A discrete multitone modem according to claim 188 which said at least one parameter comprises a signal to noise ratio of communications over said line.

190. A discrete multitone modem according to claim 188 which said at least one parameter comprises a characteristic of a monitor tone transmited over said line.

191. A discrete multitone modem according to claim 188 which said characteristic comprises at least the amplitude of said tone.

192. A discrete multitone modem according to claim 188 which said characteristic comprises at least the phase of said tone.

193. A discrete multitone modem according to claim 188 which said characteristic comprises at least the frequency of said tone.

194. A discrete multitone modem according to claim 189 in which said tone is transmitted over over at least one subchannel intermediate an upstream and a downstream set of data subchannels over which said modem communicates.

195. A discrete multitone modem according to claim 189 in which said signal to noise ratio is based on measurements of reference frames transmitted over said line.

196. A discrete multitone modem according to claim 26 in which said signal to noise ratio is based on measurements of data transmitted over said line.

197. A discrete multitone modem according to claim 177 in which the means responsive to a disturbance event comprises means for measuring at least one characteristic of said line indicative of communications on said line and for selecting a power level responsive to said measurement.

198. A discrete multitone modem according to claim 197 in which said characteristic comprises CRC errors and in which said measuring means signals a change in power level when said CRC errors exceed a defined threshold on a selected plurality of successive measurements thereof.

199. A discrete multitone modem according to claim 197 in which said characteristic comprises forward error correction coefficients and in which said measuring means signals a change in power level when the number of errors exceeds a defined threshold.

200. A discrete multitone modem according to claim 199 in which said measuring means signals a change in power level when the number of uncorrected errors exceeds a defined threshold.

201. A discrete multitone modem according to claim 199 in which the means for communicating the change in power level designates a single alternative power level to which the remote receiver is to switch.

202. A discrete multitone modem according to claim 177 which includes means in said modem for at least one parameter indicative of communication

203. A method of transmitting data over a wire line through upstream and downstream channels, respectively, from first and second pluralities of discrete-frequency subchannels, comprising the steps of:

A. storing at least first and second parameter sets defining data communications over said channels under at least two different communication conditions;

B. selecting a parameter set for use in communications in accordance with the prevailing communication condition.

204. The method of claim 203 in which said selecting step includes the step of monitoring communications on said line and transmitting and selecting said parameter set in accordance with said monitoring.

205. The method of claim 204 in which said monitoring step includes the step of measuring at least one communication indicium on said at least one subchannel.

206. The method of claim 205 in which said at least one indicium is selected from the group comprising signal to noise ratios, error rates, and the amplitude and frequency of tones.

207. The method of claims 203 or 206 which includes the step of transmitting over said line a signal that identifies the parameter set to be selected.

208. The method of claims 203 or 206 which includes the step of receiving over said line a signal that identifies the parameter set to be selected.

209. The method of claim 207 in which said signal is transmitted on a subchannel intermediate said upstream and downstream channels.

210. The method of claim 208 in which said signal is received on a subchannel intermediate said upstream and downstream channels.

211. The method of claims 203, 206 or 207 in which said first parameter set defines communications over said line in the absence of a disturbance event and said second parameter set defines communications over said line in the presence of a disturbance event.

212. The method of claims 203 or 211 in which said parameter sets include at least one parameter set from the group comprising subchannel bit allocations subcahnel gains.

213. The method of claims 203 or 211 in which said parameter sets include at least one parameter set from the group comprising subchannel frequency domain coefficients, time domain coefficients, and echo cancellation coefficients.

214. The method of claims 212 or 213 in which said parameter sets include a first section for use in transmitting data over said line and a second portion for receiving data over said line.

215. A method of transmitting data over a wire line through upstream and downstream channels, respectively, from first and second pluralities of discrete-frequency subchannels, comprising the steps of:

A. signaling over said line to a remote receiver the intention to transmit data over said line at a selected one of a plurality of predefined power levels;

B. transmitting data over said line at said selected power level

216. The method of claims 214 or 219 which includes the step of monitoring communications conditions on said line and selecting said power level in accordance therewith.

217. The method of claims 215 or 219 in which the step of selecting said power level includes the step of selecting a first power level in response to detecting the absence of a disturbance event and selecting a second power level in response to detecting the presence of a disturbance event.

218. The method f claim 217 in which said second power level is selected from a group of at least two power levels.

219. A method of transmitting data over a wire line through upstream and downstream channels, respectively, from first and second pluralities of discrete-frequency subchannels, comprising the steps of:

A. signaling to a remote receiver at one of a plurality of power levels;

B. receiving a signal from a receiver that determines said power levels.

220. The method of claim 219 in which said power levels are selected from a plurality of predetermined power levels having corresponding pre-stored parameter sets.

221. The method of claim 219 in which said power levels are received via said signal from said remote receiver.

222. The method of claim 219 in which said signal includes at least one signal selected from the group comprising a message, a tone, a collection of tones, or a flag.

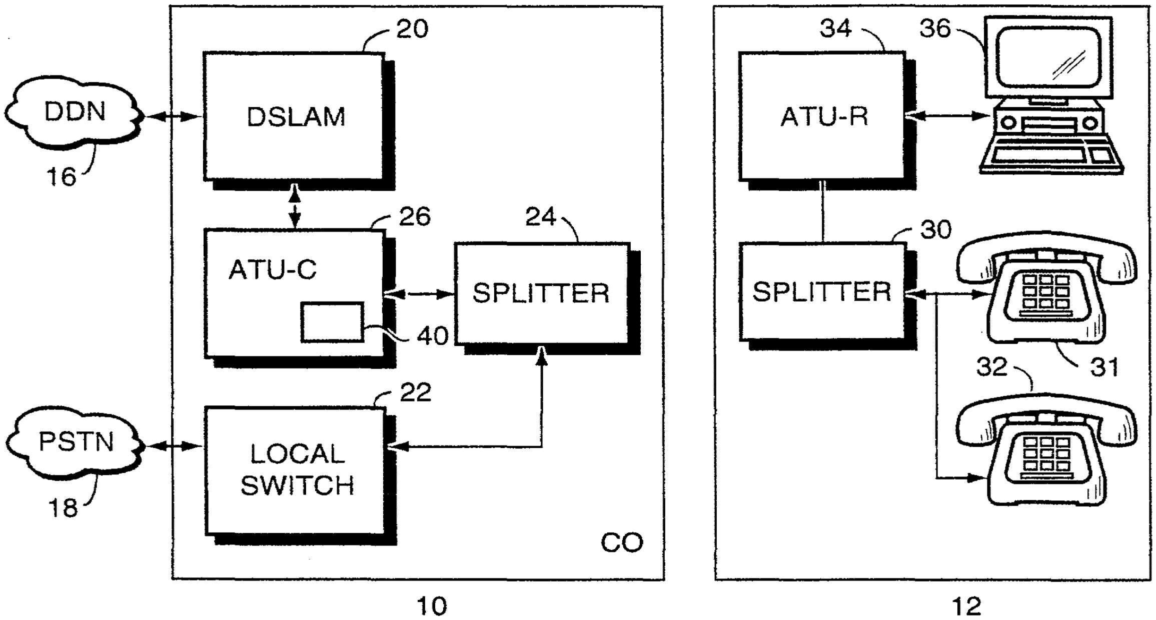


FIG. 1 (PRIOR ART)

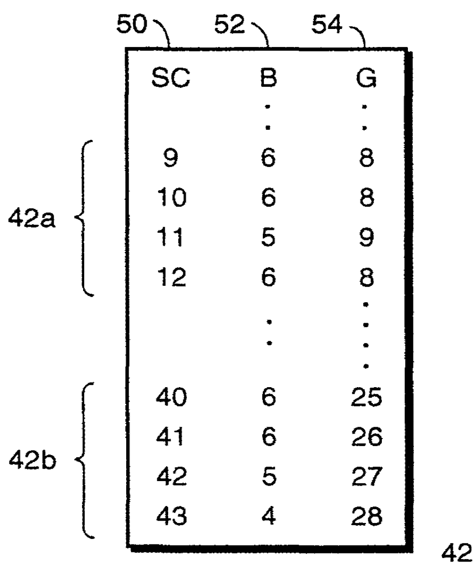


FIG. 2 (PRIOR ART)

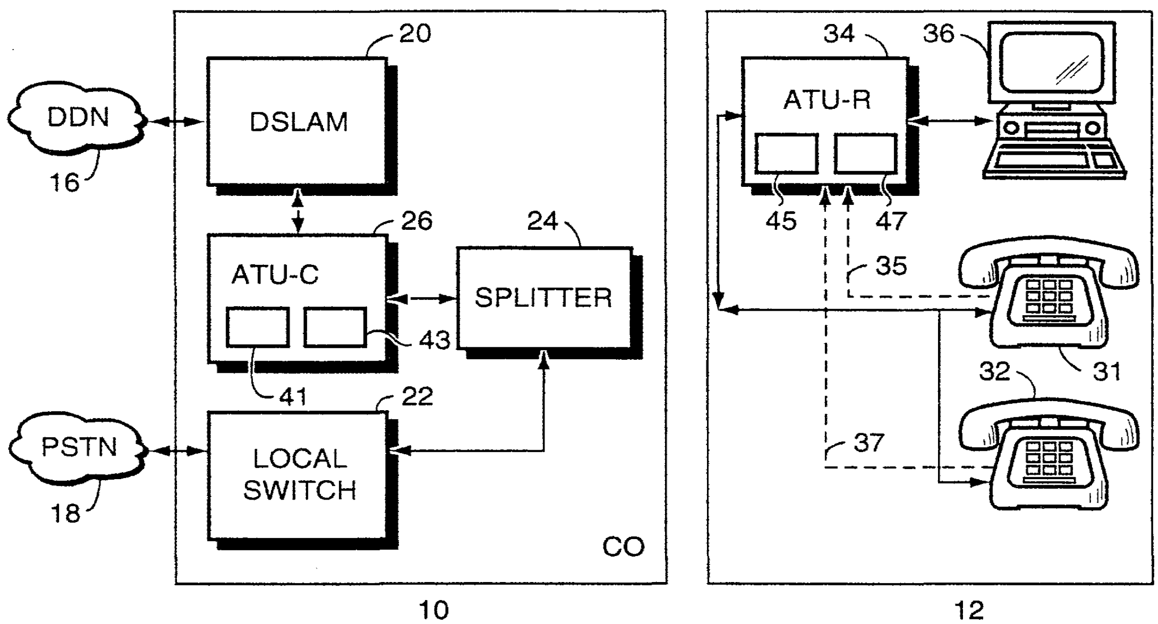

DDN
16
DSLAM
20
26
ATU-C
41
43
24
SPLITTER
22
LOCAL
SWITCH
PSTN
18
CO
34
36
ATU-R
45
47
35
31
32
37
10
12

FIG. 3

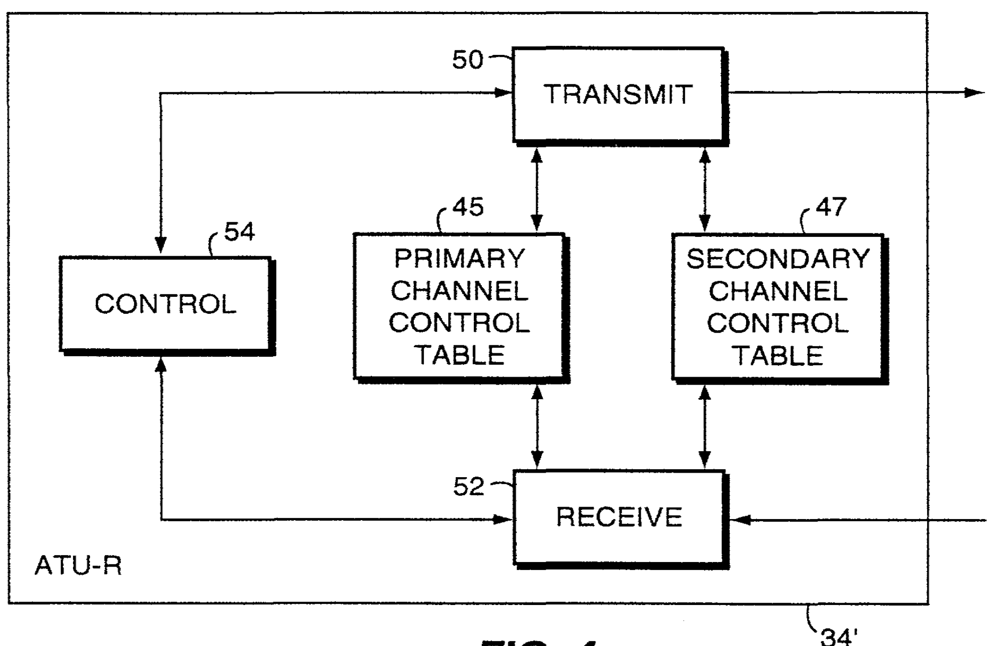

50
TRANSMIT
54
CONTROL
45
PRIMARY
CHANNEL
CONTROL
TABLE
47
SECONDARY
CHANNEL
CONTROL
TABLE
52
RECEIVE
ATU-R
34'

FIG. 4

45c 45d 45e 45f 45g 45h

	SC	B	G	FDQ	TDQ	EC
45a	9	8	0			
	10	8	0			
	11	7	1			
	12	8	0			
45b	40	7	1	a	c	e
	41	7	1	a	c	e
	42	7	1	a	c	e
	43	6	1.3	b	d	f
	:	:	:	:	:	:

FIG. 5A

47a

47f 47g 47h 47i 47j 47b 47c 47c

	SC	B	G	FDQ	TDQ	EC	B	G	FDQ	TDQ	EC	B	G	FDQ	TDQ	EC
47d	9	7	1				8	-8				8	-8			
	10	6	1.3				8	-8				8	-8			
	11	7	1				7	-8				7	-8			
	12	7	1				8	-8				8	-8			
47e	40	7	1	g	i	k	7	1	m	p	s	6	1.3	u	x	t
	41	7	1	g	j	k	6	1.3	n	q	t	4	1.6	v	y	1
	42	7	1	h	i	k	5	1.5	o	r	u	5	1.5	w	z	r
	43	6	1.3	g	i	l	6	1	n	q	t	6	1.3	v	x	t
	:	:	:	:	:	:	:	:	:	:	:	:	:	:	:	:

47

FIG. 5B

SUBSTITUTE SHEET (RULE 26)

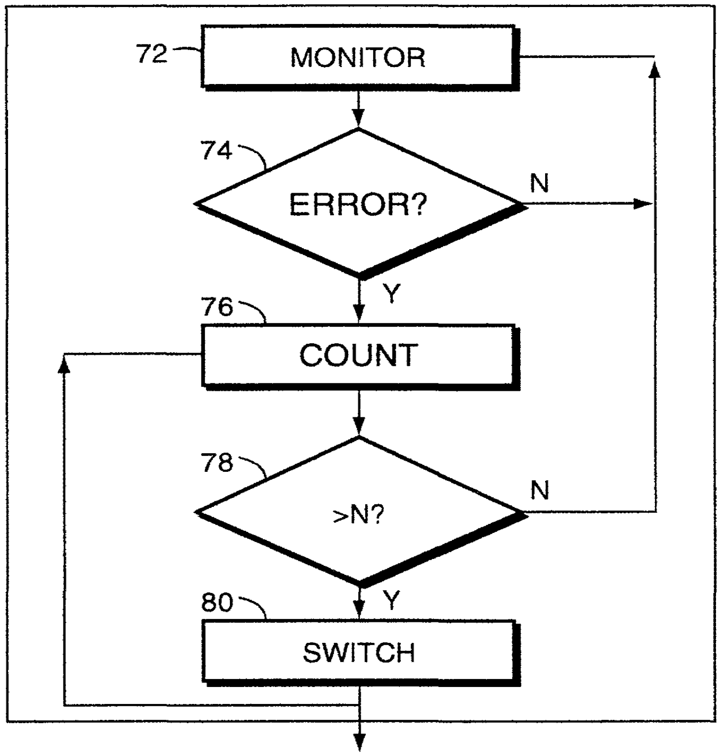


FIG. 6

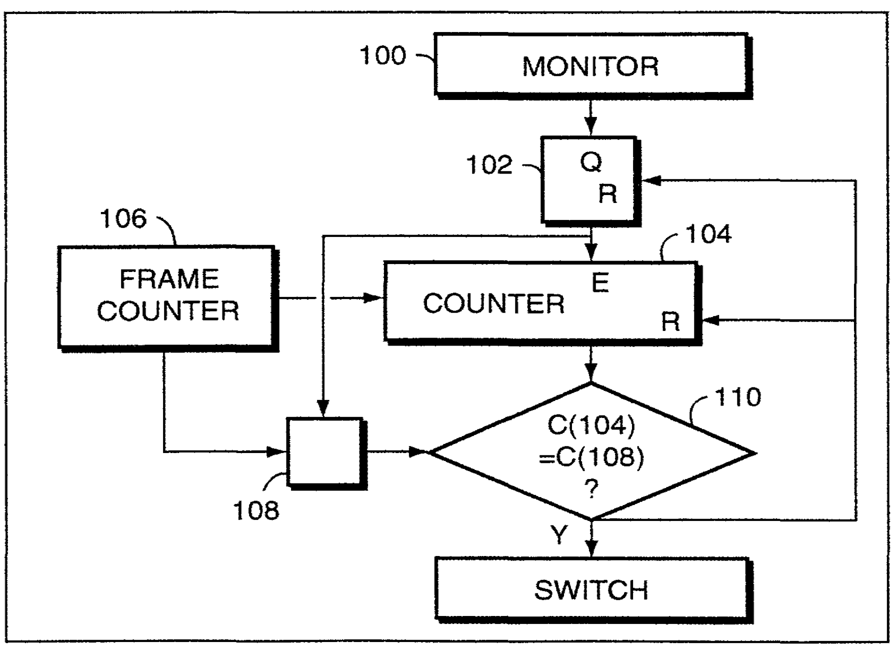
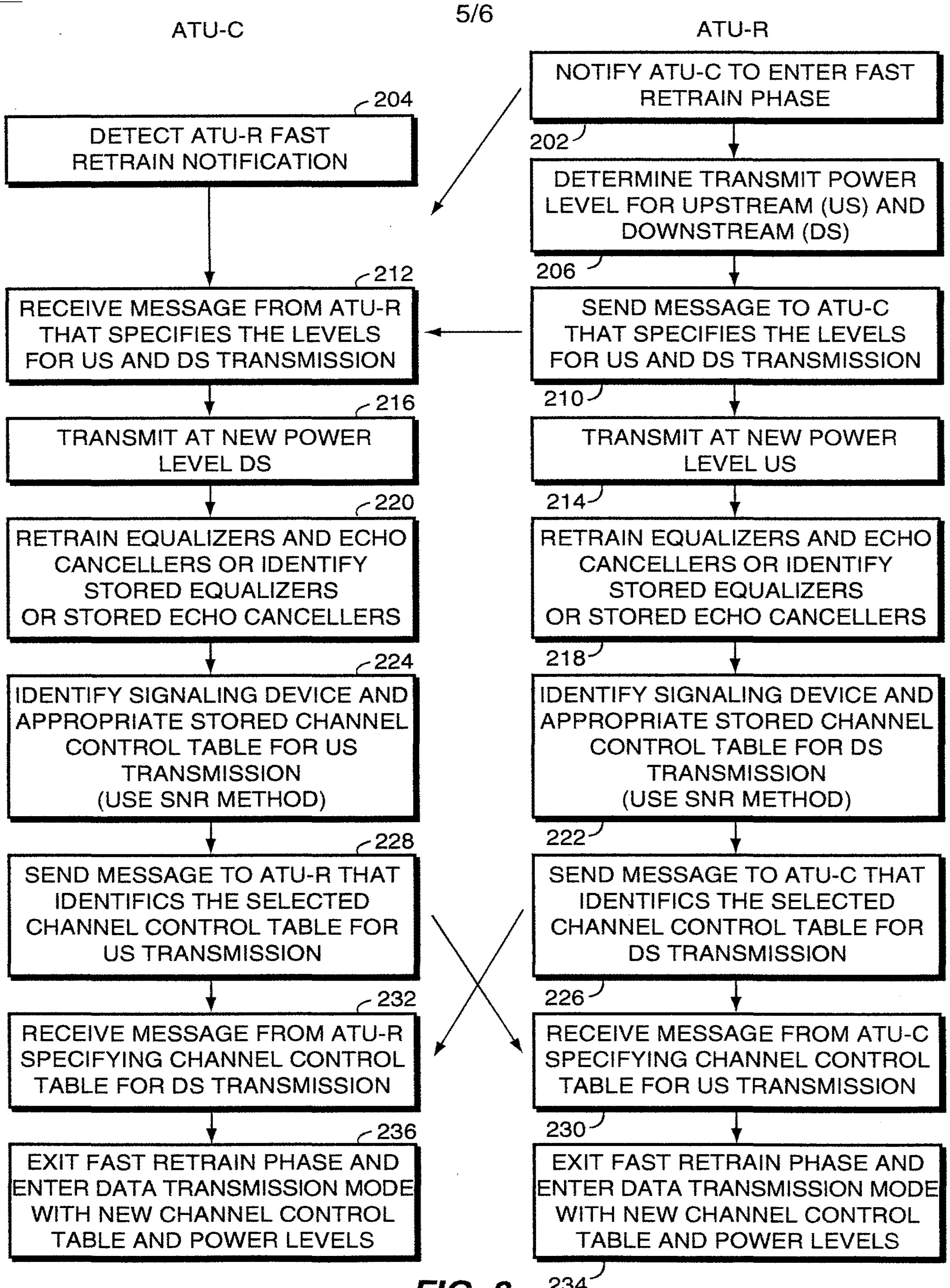


FIG. 7

SUBSTITUTE SHEET (RULE 26)



ATU-C
ATU-R
NOTIFY ATU-C TO ENTER FAST RETRAIN PHASE
202
204
DETECT ATU-R FAST RETRAIN NOTIFICATION
DETERMINE TRANSMIT POWER LEVEL FOR UPSTREAM (US) AND DOWNSTREAM (DS)
206
212
RECEIVE MESSAGE FROM ATU-R THAT SPECIFIES THE LEVELS FOR US AND DS TRANSMISSION
SEND MESSAGE TO ATU-C THAT SPECIFIES THE LEVELS FOR US AND DS TRANSMISSION
210
216
TRANSMIT AT NEW POWER LEVEL DS
TRANSMIT AT NEW POWER LEVEL US
214
220
RETRAIN EQUALIZERS AND ECHO CANCELLERS OR IDENTIFY STORED EQUALIZERS OR STORED ECHO CANCELLERS
RETRAIN EQUALIZERS AND ECHO CANCELLERS OR IDENTIFY STORED EQUALIZERS OR STORED ECHO CANCELLERS
218
224
IDENTIFY SIGNALING DEVICE AND APPROPRIATE STORED CHANNEL CONTROL TABLE FOR US TRANSMISSION (USE SNR METHOD)
IDENTIFY SIGNALING DEVICE AND APPROPRIATE STORED CHANNEL CONTROL TABLE FOR DS TRANSMISSION (USE SNR METHOD)
222
228
SEND MESSAGE TO ATU-R THAT IDENTIFICS THE SELECTED CHANNEL CONTROL TABLE FOR US TRANSMISSION
SEND MESSAGE TO ATU-C THAT IDENTIFICS THE SELECTED CHANNEL CONTROL TABLE FOR DS TRANSMISSION
226
232
RECEIVE MESSAGE FROM ATU-R SPECIFYING CHANNEL CONTROL TABLE FOR DS TRANSMISSION
RECEIVE MESSAGE FROM ATU-C SPECIFYING CHANNEL CONTROL TABLE FOR US TRANSMISSION
230
236
EXIT FAST RETRAIN PHASE AND ENTER DATA TRANSMISSION MODE WITH NEW CHANNEL CONTROL TABLE AND POWER LEVELS
EXIT FAST RETRAIN PHASE AND ENTER DATA TRANSMISSION MODE WITH NEW CHANNEL CONTROL TABLE AND POWER LEVELS
234

FIG. 8

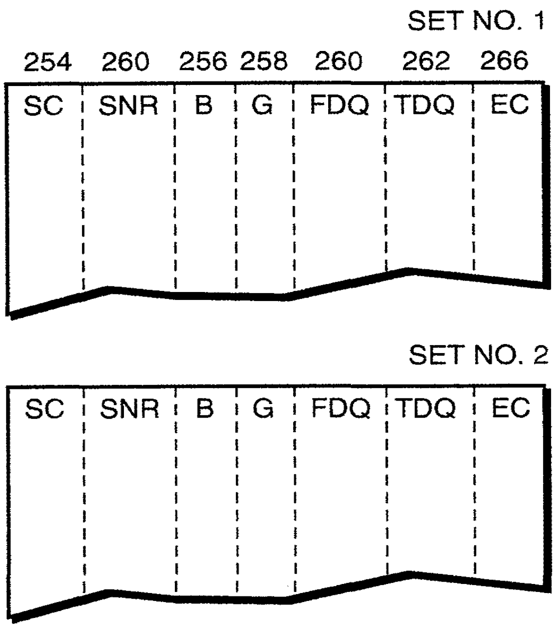


FIG. 9A

49e 49a 49b 49c 49d 49f

SC	B1	B2	B3	B4	B5	B'
9	8	7	7	6	5	5
10	8	6	6	4	4	4
11	7	7	6	5	5	5
12	8	7	6	5	5	5
40	7	7	6	6	5	5
41	7	7	6	5	5	5
42	7	7	6	5	5	5
43	6	7	6	5	5	5
:	:	:	:	:	:	:

FIG. 5C

270 272 274 276

{SET No., SNR1, SNR2, SNR3, ...}

FIG. 9B

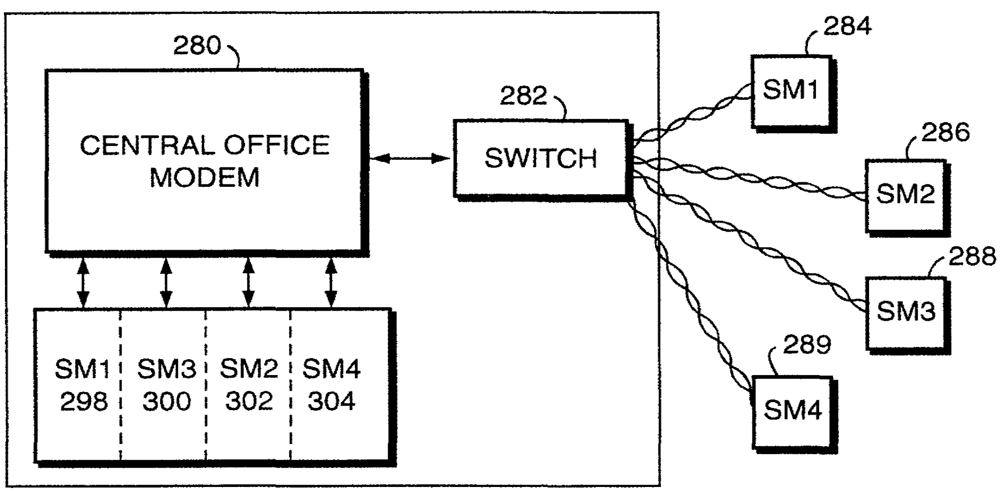


FIG. 10

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INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 :
H04L 12/00
A2

(11) International Publication Number: **WO 99/26375**

(43) International Publication Date: 27 May 1999 (27.05.99)

(21) International Application Number: PCT/US98/23336

(22) International Filing Date: 2 November 1998 (02.11.98)

(30) Priority Data:
60/066,021 14 November 1997 (14.11.97) US

(71) Applicant *(for all designated States except US)*: TEKTRONIX, INC. [US/US]; 14150 S.W. Karl Braun Drive, Mail Stop 50–LAW, P.O. Box 500, Beaverton, OR 97077 (US).

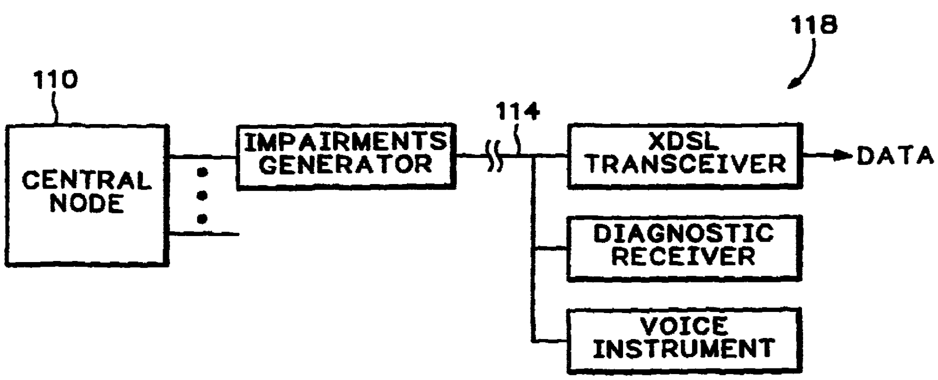
(72) Inventor; and
(75) Inventor/Applicant *(for US only)*: HAKANSON, Eric, W. [US/US]; 12190 N.W. Welsh Drive, Portland, OR 97229 (US).

(74) Agent: BUCHER, William, K.; Tektronix, Inc., Mail Stop 50–LAW, P.O. Box 500, Beaverton, OR 97077 (US).

(81) Designated States: CA, CN, KR, US, European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

Published
Without international search report and to be republished upon receipt of that report.

(54) Title: METHOD OF OPERATING A DIGITAL DATA DISTRIBUTION NETWORK



(57) Abstract

Digital data in error protected packets is used to modulate a carrier and the modulated carrier is impressed on a digital data distribution network for transmission to a receiver over a transmission path. In order to monitor operation of the data distribution network, the transmission path is impaired to a selected extent upstream of a transmission path segment that is to be tested and an error protected data packet is transmitted over the transmission path to the receiver. A determination is made at the receiver whether the received data packet is error free, and, if not, a message is transmitted from the receiver to the transmitter.

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METHOD OF OPERATING A DIGITAL DATA DISTRIBUTION NETWORK

Background of the Invention

This invention relates to method of operating a digital data distribution network.

In a conventional cable television system, a video information signal in analog form, such as the NTSC composite video signal, is employed to modulate the RF carrier of an assigned RF transmission frequency channel at the system headend and the RF signal is distributed over a cable network to multiple subscriber nodes. At a subscriber node there may be a cable-ready television receiver including a tuner which can select the frequency channel and a detector which recovers the video information signal from the selected channel and employs it to control operation of the television display.

A data distribution system in which the information signal is transmitted in digital form has well known advantages over a system in which the information signal is transmitted in analog form. Accordingly, it has been proposed by the United States Federal Communications Commission (FCC) that terrestrial transmission systems under the jurisdiction of the FCC should phase out use of the NTSC composite video signal by 2007 and should instead use digital video information signals to modulate RF carriers. The digital video information signal provided by a video signal source will then be composed of a succession of bits segregated into digital data packets. The data packets modulate an RF carrier which is broadcast from the transmitter. Each period of the RF carrier conveys several bits of the digital information signal in one symbol. For example, in the 64QAM modulation scheme, each symbol conveys six bits of the digital information signal. The television receiver selects the frequency channel, detects an analog information signal, converts the detected information signal to digital form and recovers the digital data packets. The digital information signal is then used to control operation of the television display.

The change in standards from analog to digital for terrestrial television transmission effectively dictates that cable television systems will also have to provide digital video signals in order for the video signals to be compatible with digital television receivers.

Referring to FIG. 1, a digital cable television system includes a digital processing interface 8 which receives a digital video information signal, such as the MPEG transport stream, and generates an error protected digital signal composed of a succession of error protected digital signal packets. The error protected digital signal is applied to a modulator 10 which employs it to modulate an RF carrier which is typically in the frequency range 50-550 MHz, although it may be higher or lower. The digitally modulated RF carrier is supplied to a transmitter 14 which impresses the signal on a propagation medium 16. In the case of a cable television system, the propagation medium is a network of coaxial cables configured as a trunk extending from the transmitter 14 and having numerous branches connected to the trunk by directional couplers 18, sub-branches connected to the branches by directional couplers, and so on, and connected at the subscriber nodes to digital television receivers 20.

Each receiver 20 has a front end 22 including a tuner (not shown) which converts the RF signal to intermediate frequency and an analog-to-digital converter (ADC) 26 which digitizes the IF signal and provides a digital output signal to a demodulator 30. The demodulator 30 removes the IF component and provides a digital output signal, which, ideally, should match the error protected digital signal provided to the modulator 10. The receiver front end 22 also includes a digital processing circuit 32 which carries out the inverse of the error protection algorithm employed at the headend and ideally provides at its output a digital video information signal which matches the signal supplied to the digital processing interface 8. The digital video information signal from the digital processing circuit 32 is supplied through a decoder (not shown) which the MPEG transport stream and supplies an analog video signal to display circuitry 34 to control operation of the television display.

Error protection is employed in the digital cable television system to allow correction of bit errors, i.e. incorrect values of digital 1 or digital 0, in the output signal of the demodulator 30 caused by impairments in the transmission path from the input of the modulator 10 to the output of the demodulator 30.

Provided that the bit error rate is below a critical value, known as the critical bit error rate and generally considered to be about 10^{-4} for a digital television signal, digital error correction techniques can correct the errors and provide a signal having a bit error rate that may be less than 10^{-11}, which is sometimes referred to as quasi-error free. The maximum bit error rate that can be tolerated is considered to be about 10^{-3} before error correction.

Some video signals in a cable system are transmitted in encrypted form in order to restrict their use to subscribers who have paid an additional fee, either on a periodic basis for premium channels or on a pay-per-view basis for particular programs. In this case, the digital processing interface 8 not only applies a digital error protection algorithm but also encrypts the digital video information signal, so that the digital data packets provided to the modulator 10 are error protected and encrypted. In order to decrypt the digital data packets and regenerate the analog video signal, the subscriber is provided with a set top terminal 40 which is connected between the cable system connection and the display circuitry 34, by-passing the front end 22. The set top terminal includes a tuner (not shown) an ADC 42, a demodulator 44 and a digital processing circuit 46, performing the same general functions as the front end 22, but the digital processing circuit 46 performs not only error correction to recreate the digital signal applied to the modulator 10 but also decryption in order to extract the digital video information signal supplied to the digital processing interface 8. It is expected that much of the programming distributed by digital transmission cable systems will be transmitted in encrypted form, so that a subscriber will need a set top terminal, or equivalent functionality built into the television receiver, in order to display a variety of programming.

The economic value of a cable television distribution system resides in its ability to distribute video payload, i.e. the program material that subscribers wish to view, to a large number of subscribers without excessive degradation. The system operator derives revenue based on the system's ability to distribute the video payload. Accordingly, it is important that the system operator be warned of impairments in the distribution system, so that these impairments can be corrected before they adversely affect the ability of the system to distribute video payload and hence the revenue derived by the system operator. The operator must therefore be able to measure impairments in transmission quality so that appropriate repairs can be made. Typical impairments that should be detected and repaired are reductions in signal-to-noise ratio (SNR), e.g. due to noise being coupled into the transmission channel, reductions in frequency response, reductions in phase response, phase noise, jitter, addition of interfering signals and addition of multipath signals.

Hitherto, it has been suggested that the bit error rate of an RF digital transmission system may be a satisfactory measure of transmission channel quality, but this measure is subject to disadvantage because an RF data distribution system in which the information signal is digital is subject to the "cliff effect," in that the curve that relates bit error rate to the quality of the transmission channel, expressed as signal-to-noise ratio, has a very steep drop off. Thus, referring to FIG. 2, a change of less than 1.5 dB in signal-to-noise ratio can cause the bit error rate to change from less than 10^{-4} to more than 10^{-3}. The curves shown in FIG. 2 assume that the only impairment is noise when in fact there will always be other impairments, which can make the drop off even steeper. Accordingly, the system operator is not alerted to impairment of the transmission quality of the channel either by BER measurements or by a relatively small increase in subscriber complaints. On the contrary, the operator may not learn of an impairment until the system fails. This makes it difficult to monitor the noise margin in the system, to track degradations and fix degradations before a system failure.

In a report issued by the European Telecommunications Standards Institute (ETR 290: May 1997), it is suggested that the estimated noise margin is a better indicator of transmission channel quality than bit error rate. The estimated noise margin is based on the probability of mathematically added noise causing a bit error and is approximately the difference between the current estimated signal-to-noise ratio and the estimated signal-to-noise ratio at which the bit error rate exceeds the critical bit error rate. Use of the estimated noise margin to identify impairments is subject to disadvantage because it is computationally expensive and is not applicable to impairments other than noise. Further, its reliability is limited because there is an unknown set of errors associated with calculating the estimated noise margin. Since the estimated noise margin is not the same as the actual noise margin, there is a possibility that the current signal-to-noise ratio is substantially less than the estimated signal-to-noise ratio, and consequently the actual noise margin may be substantially less than the estimated noise margin. It would therefore be desirable to determine the actual noise margin of the transmission channel.

Summary of the Invention

In accordance with a first aspect of the invention there is provided a method of operating a digital data distribution network having a transmitter and a receiver, wherein digital data is transmitted in error protected packets from the transmitter to the receiver over a transmission path by employing the digital data to modulate at least one carrier and impressing the modulated carrier on the network, said method comprising (a) generating an error protected data packet for transmission over the transmission path, (b) impairing the transmission path to a selected extent upstream of a transmission path segment that is to be tested, (c) transmitting the data packet over the transmission path, (d) receiving the data packet at the receiver, and (e) determining whether the received data packet is error free.

In accordance with a second aspect of the invention there is provided a method of operating a digital data distribution network having a transmitter and a receiver, wherein digital data is transmitted in error protected packets from the transmitter to the receiver over a transmission path by employing the digital data to modulate at least one carrier and impressing the modulated carrier on the network, said method comprising (a) generating an error protected data packet for transmission over the transmission path, (b) transmitting the data packet over the transmission path as an analog signal, (c) receiving the analog signal at the receiver, (d) recording the analog signal received at the receiver, and (e) transmitting the record of the analog signal to a remote location for analysis.

In accordance with a third aspect of the invention there is provided a method of operating a digital data distribution network having a transmitter and a receiver, wherein digital data is transmitted in error protected packets from the transmitter to the receiver over a transmission path by employing the digital data to modulate at least one carrier and impressing the modulated carrier on the network, said method comprising generating an error protected data packet for transmission over the transmission path, impairing the transmission path to a selected extent upstream of a transmission path segment that is to be tested, transmitting the data packet over the transmission path, receiving the data packet at the receiver, and counting bit errors in the received data packet.

In accordance with a fourth aspect of the invention there is provided a method of operating a digital data distribution network having a transmitter and a receiver, wherein digital data is transmitted in error protected packets from the transmitter to the receiver over a transmission path by employing the digital data to modulate a carrier and impressing the modulated carrier on the network, said method comprising (a) generating an error protected data packet for transmission over the transmission path, (b) impairing the transmission path to a selected extent upstream of a transmission path segment that is to be tested, (c) transmitting the data packet over the transmission path as an analog signal, and (d) receiving the analog signal at the receiver.

Brief Description of the Drawings

For a better understanding of the invention, and to show how the same may be carried into effect, reference will now be made, by way of example, to the accompanying drawings, in which

FIG. 1 is a partial schematic block diagram of a proposed form of cable television system,

FIG. 2 is a graph illustrating bit error rate as a function of signal-to-noise ratio in a digital data communication system.

FIG. 3 is a partial schematic block diagram of the headend and receiver in a cable television system embodying the present invention,

FIG. 4 is a map of part of a cable television system, and

FIG. 5 is a partial schematic block diagram of a digital subscriber line system.

Detailed Description

A first application of the invention will be described with reference to a digital cable television system.

The cable TV system shown in FIGS. 3 and 4 is used to distribute digital video information signals from a headend 48 to subscriber nodes 50. Referring to FIG. 3, the headend 48 includes a digital processing interface 8, a modulator 10 and a transmitter 14 similar to the corresponding elements shown in FIG. 1. The digital processing interface receives the MPEG 2 transport stream and performs various operations, including energy dispersal, error protection, interleaving and base band shaping in order to generate inphase and quadrature signals which are applied to the modulator 10. All the functions of the digital processing interface, and possibly also the functions of the modulator, may be performed in a single integrated circuit. In addition, the headend 48 includes an impairments generator 60. The impairments generator 60 may be located between the transmitter 14 and the cable network 16, as shown in FIG. 3, or it may be incorporated in the digital processing interface 8 or the modulator 10. The effect of the impairments generator 60 is to degrade to a selectively controllable extent the quality of the transmission path between the digital processing interface 8 and the subscriber nodes 50. The impairments generator may function by adding noise to the transmission channel or degrading the frequency response or phase response of the transmission channel. Further, the impairments generator may introduce "spurs" (spurious modulation products) and phase noise or jitter. The manner in which the impairments can be applied to the transmission channel is well known to those skilled in the art. Considering, for example, the signal-to-noise ratio, the quality of the channel may be degraded at the headend using an impairments generator that couples noise into the transmission channel. The extent to which the signal-to-noise ratio is degraded depends on the amplitude of the noise.

As shown in FIG. 4, the cable network 16 includes a trunk extending from the headend 48. Branches and sub-branches are connected to the trunk by directional couplers 18. Each subscriber node 50 is at the end of a branch or sub-branch. The cable system operator maintains a map of the cable network, showing schematically the topology of the path to each subscriber node 50. At each active subscriber node 50, there is a diagnostic cable receiver 64 (FIG. 3) connected between the cable network and the display circuitry 34 of the subscriber's digital television receiver. The cable receiver 64 may be implemented as a set top terminal or it may be housed in the same cabinet as the digital television receiver. Each cable receiver has a unique ID and the cable system operator maintains a database relating cable receiver IDs with the subscriber nodes and billing addresses. If the database also relates the cable receiver IDs with physical addresses, the system operator is able to determine not only the physical location of each cable receiver but also the topology of the path between the headend and each cable receiver.

Referring to FIG. 3, the cable receiver 64 includes a tuner (not shown) for converting the received signal to the intermediate frequency, an ADC 66, a demodulator 68, a digital processing circuit 70 and a decoder (not shown), similarly to the set top terminal 40 described with reference to FIG. 1. A controller 74 included in the cable receiver controls operation of the other components of the cable receiver 64.

The capabilities of the digital processing circuit 70 are expanded relative to those of the digital processing circuit 46. The digital processing circuit 70 has a video data output for supplying the MPEG transport stream to the decoder, which supplies an analog video signal to the display circuitry 34 of the digital television receiver. The digital processing circuit 70 includes an error bits counter which accumulates the number of error bits in the received signal. The error bits counter can be queried by the controller 74 and reset from time to time, so that the controller is able to calculate the bit error rate based on the error bit count and the time that has elapsed since the counter was reset. The controller 74 supplies a digital data word representing the calculated value of the bit error rate to a digital processing interface 76, which produces an error protected data packet.

The cable receiver 64 also includes a memory 80 which can be enabled to store the output signal of the ADC 66 during a selected interval. The stored digital signal is applied to the digital processing interface 76 to generate an error protected data packet. The error protected data packet produced by the digital processing circuit 76, either from the bit error rate word or from the signal provided by the memory 80, is supplied to a modulator 82. The modulator 82 uses the error protected data packet to modulate an RF carrier, typically at a frequency in the range 5-50 MHz, although it may be higher or lower. The modulated RF signal is applied to a transmitter 84 which impresses the signal on the cable network.

The headend 48 of the cable system also includes a receiver 90 for receiving the return messages provided by the transmitter 84 in each of the cable receivers 64. The receiver 90 includes a tuner (not shown), an ADC 92 which digitizes the return message signal, a demodulator 94 which removes the IF component and provides a digital output signal which, ideally, should match the error protected return message packet provided by the digital processing interface 76, a digital processing circuit 96 which carries out the inverse of the error protection algorithm employed in the digital processing interface 76 and ideally provides at its output a data signal which matches the input signal provided to the digital processing interface 76, and a report/display device 98. It will be understood that the headend includes a controller (not shown) for controlling operation of the various components thereof.

In a first mode of operation of the cable television system shown in FIG. 3, the system is used to measure the bit error rate of the transmission channel to each of the subscriber nodes. In this mode of operation, the headend controller issues a signal which is transmitted to the cable receivers, instructing the cable receivers to calculate bit error rate during a selected measurement interval, which may be defined by reference to start and stop flags included in the data stream or by reference to specific start and stop times supplied to the cable receivers by the headend.

During the measurement interval, the controller 74 calculates the bit error rate and provides an output word representative thereof. The calculated bit error rate is reported back to the headend with the cable receiver ID and a report or display is generated. The report/display device may accumulate information received from numerous cable receivers 64 and generate a report or display showing trends in bit error rate with time.

Alternatively, or in addition, the report/display device may generate a report or display showing bit error rate as a function of the locations of the cable receivers in the cable network, for example. The system operator is thereby able to determine, on a node-by-node basis, the bit error rates of the signal propagation paths between the transmitter 14 and the subscriber nodes. By comparing the bit error rates reported by different cable receivers, the cable system operator may be able to determine the location in the cable network of a particular impairment. For example, referring to FIG. 4, if the cable receivers at nodes 50C and 50D have poor transmission margin compared to the terminals at nodes 50A, 50B, 50E and 50F, indicated by high bit error rate, then it is likely that there is an impairment between the directional couplers 18_2 and $18_{2,1}$.

It will be appreciated that a test of this nature will generate a response message from each cable receiver, and accordingly it may be advantageous to instruct only selected cable receivers to calculate the bit error rate and provide return messages.

As noted previously, the bit error rate of the propagation path may be of limited value for monitoring degradation of the transmission quality, and it may be better to measure noise margin.

In order to measure the noise margin, i.e. the difference between the current signal-to-noise ratio and the SNR at which the bit error rate exceeds the critical bit error rate, the headend controller instructs the cable receivers (or a selected group of cable receivers) to report when the bit error rate calculated by the controller 74 exceeds the critical bit error rate. The headend controller operates the impairments generator 60 to add a noise impairment to the signal emitted by the transmitter. The noise amplitude is progressively increased, for example in stair-step fashion. In each of the cable receivers addressed by the headend controller, the controller 74 provides an output indicating the bit error rate. When the bit error rate at a given cable receiver 64 without addition of the noise impairment is sufficiently low, and the bit error rate with addition of the noise impairment exceeds the critical bit error rate, the level of impairment introduced by the impairments generator is approximately equal to the noise margin for the transmission channel from the transmitter 14 to that cable receiver. (If the impairments generator were upstream of the transmitter, the level of impairment introduced by the impairments generator would be related to the noise margin for the segment of the transmission path between the impairments generator and the cable receiver.) The cable receiver reports that the critical bit error rate has been exceeded, and includes its ID in the report. The cable system operator is thereby able to determine the noise margin to critical bit error rate on a node-by-node basis by correlating the cable receiver IDs with the level of impairment at which each cable receiver provides a report. It is, of course, necessary to correlate the report that the critical bit error rate has been exceeded with the noise level at which the report was generated. This may be accomplished by including framing bits in the signal transmitted by the head end in the event that the cable receiver reports immediately that the critical bit error rate has been exceeded. Alternatively, the headend controller may maintain a log recording level of impairment as a function of time and the report could include a time stamp indicating the time at which the critical bit error rate was exceeded.

It may be helpful in locating system impairments in the system shown in FIG. 3, to apply an impairment to the transmission channel and observe the effect of that impairment at multiple locations simultaneously.

If there is an impairment in the trunk of the cable network or in a major branch, it is likely that many cable receivers will respond to the stair-step type of impairment and the reverse transmission system would become jammed by the message storm. This can be avoided by testing all cable receivers at relatively short intervals, with a small level of impairment. Appropriate selection of the level of impairment should ensure that relatively few cable receivers will report a malfunction or failure condition. If this indeed occurs, the operator then has confidence that the transmission channel has a reasonable margin. If there is an unexpectedly large number of return messages, the headend controller may broadcast a message to all cable receivers instructing them not to send error information but to reset and measure again. The headend then repeats the test with a lower level of impairment in order to locate the regions of the network for which the noise margin is smallest. At longer intervals, e.g. daily or monthly, the operator tests all cable receivers with a stair-step sequence of impairments preceded by a message that the cable receivers should report the result of the test only when polled. The headend then polls the cable receivers and the cable receivers respond to the poll by reporting the actual transmission margin. The polling is best done during an idle period, so as not to interfere with revenue generating transmissions. Since the transmission margin from the headend to each cable receiver can be inexpensively monitored, the problem of locating an impairment in the cable network is greatly simplified.

If multiple impairments exist, it can be difficult to locate the impairment responsible for a failure condition. For example, referring to FIG. 4, the reduction in transmission margin downstream of an impairment in cable segment 102 may be quite small and may be swamped by another impairment upstream in the system, e.g. in cable segment 104. Alternatively, two different impairments, e.g. in cable segments 102 and 106, may cause similar reductions in transmission margin, thus leading to the erroneous conclusion that there is a single impairment in a branch that is common to the nodes 50C and 50E, e.g. cable segment 104. If the impairments are of different types, e.g. noise and jitter, this problem can be solved by classifying the impairments.

In order to classify impairments, it is necessary to observe the effect of the impairments on symbols, as opposed to the bits used to encode the symbols.

Impairments can be classified by comparing the waveform of the signal received at the subscriber node with the waveform of the transmitted signal.

This is accomplished by using the memory 80 to capture the digital output signal of the ADC 66 during a test interval and transmitting the captured waveform back to the headend. The digital processing circuit 96 provides an output signal that matches the captured portion of the output signal of the ADC 66 and can be compared with the output signal of the transmitter 14 during the corresponding time interval, so that the effect of the impairments on symbols can be determined.

Alternatively, the captured sample of the waveform can be analyzed locally using a measurement instrument.

There are several ways in which impairments can be classified. One technique is to derive the error vector waveform and extract the spectrum of the error vector. The presence of various impairments, such as noise, coherent distortions and spurious modulation products, can be deduced from the spectrum of the error vector. Amplitude and phase modulation impairments can be deduced from the Hilbert Transform of the error vector waveform.

The error vector waveform is derived by subtracting the signal received at the input of the cable receiver from the transmitted signal. Typically, the cable receiver will include an equalizer downstream of the ADC, often as part of the demodulator. If the equalizer is upstream of the point at which the received signal is read for storing in the memory 80, it affects the timing of the received signal and its effect must be removed in order for the received signal waveform to reflect the condition of the transmission path. This can be accomplished by using the equalizer coefficients to create a digital filter having a transfer function that is the inverse of the transfer function of the equalizer. The error vector waveform is then generated by subtracting the output waveform of the digital filter from the transmitted waveform.

Once the impairments have been classified, a particular existing system impairment is chosen for testing. The chosen impairment might be the impairment suspected of most likely causing a reduced transmission margin. The impairments generator then adds this impairment, at a sufficient level that the combined effect of the existing system impairment and the added impairment will be greater than the level previously detected for the existing impairment. Since the normal cable receiver is not calibrated for level, and there is a potential for destructive interference between the existing system impairment and the added impairment, the new aggregate level of impairment is best measured by repeating the recording and classification process and determining by how much the level of impairment has changed. If addition of this impairment causes a system failure report from the cable receiver at one subscriber node but not from the cable receiver at another node, it can be inferred that the two nodes are affected by different impairments.

If multiple similar impairments exist simultaneously, the impairments cannot be separated by classification and the problem of locating the impairments is more complex. However, if several of the diagnostic cable receivers are instructed to record the received waveform simultaneously, signal processing of the digitized waveforms can be used to extract the separate locations of the multiple similar impairments. The cable receivers can be made to measure simultaneously by means of two mechanisms. In accordance with the first mechanism, a protocol that instructs all cable receivers (possibly just all unused cable receivers or just selected cable receivers) to tune to a particular channel and stop recording when the end of a particular data packet is received can be broadcast to all (or some) cable receivers. This method can provide robust, but relatively coarse, timing. More precise time correlation can be achieved by inserting a time mark in the broadcast waveform, and suitable signal processing can then be used to align the received waveforms with the broadcast waveform. The time mark may be inserted by transmitting a message such that there will be a transition through a selected signal level, e.g. zero volts, at a selected time, typically late in a packet.

One way of extracting the separate locations of multiple impairments has two steps. First, the error vector waveform for each subscriber node is generated by subtracting the transmitted waveform from the waveform received at each node. Second, the cross-correlation function cev (X, Y) of the error vector waveforms for two subscriber nodes 50X and 50Y is derived. Error vectors that are common to the two nodes are revealed by the cross-correlation function. When computing the cross-correlation functions along the logical path of the network from an end point (such as a subscriber node) toward the transmitter, the location of the impairment can be determined when the value of the cross-correlation function becomes smaller. For example, referring again to

FIG. 4, and assuming that cev (E, F) indicates a common impairment and cev (A, F) indicates that the common impairment is missing, there must be an impairment between the logical locations of nodes 50A and 50E in the transmission path to node 50F. If cev (C, F) indicates a common impairment, the impairment must be between the nodes 50A and 50C. Since the only part of the network between nodes 50A and 50C that is in the transmission path to node 50F is the segment between the coupler 18_1 and the coupler 18_2, the impairment must be located there.

As a second example, if cev (C, D) indicates a common impairment and cev (A, D) indicates that the common impairment is missing, there must be an impairment logically located between node 50A and the coupler $18_{2,1}$. This implies that the impairment must be located between the directional coupler 18_1 and the coupler $18_{2,1}$. If cev (D, E) indicates that the common impairment is missing, the impairment is not between directional coupler 18_1 and the directional coupler 18_2, and so the impairment must be between the directional coupler 18_2 and the coupler $18_{2,1}$.

It is necessary to carry out the tests using the impairments generator with minimal disturbance to the revenue generating communication traffic. This is accomplished by adding the impairments only to selected packets or segments of data having a relatively low value with respect to the revenue generating communication traffic.

Most digital video transmission systems utilize the MPEG transport stream. The MPEG transport stream is composed of several MPEG elementary streams which are multiplexed to produce the MPEG transport stream. Stuffing bits are inserted in order to create the constant bit rate MPEG transport stream. It is important that the impairment should degrade only the stuffing bits or other non-customer (i.e. non-payload) bits. Since the impairments are added in the analog domain (in the case of the impairments generator being downstream of the transmitter), the impairments are applied to the transmitted symbols, in which several bits are encoded. Interleaving in constructing the transmitted data stream may result in a symbol containing bits derived from multiple elementary streams. Accordingly, it is necessary to detect when a symbol consists entirely of non-customer bits and degrade only those symbols. The cable receiver 64 can be instructed to pick out a degraded symbol by including a private data message in the MPEG transport stream. The message might, for example, instruct the cable receiver to pick out a numerically specified symbol after the next sync byte after the Program Clock Reference for a specified Program ID. It will be appreciated that the impairments could be added in the digital domain, e.g. in the digital processing interface 8. In this case, while the impairments are added in the digital domain, they are nevertheless a description of the desired analog waveform, so the impairments are of an analog nature.

An alternative is to include the impairment at a time when all of the payload bits are of relatively low perceived value. For example, the operator might include a special announcement simultaneously on all of the program streams contained in a single transmitted channel. The time of transmission of this announcement is chosen so that the balance between the loss of advertising revenue and the benefit of announcing the quality enhancement efforts is optimized. The announcement might indicate that the system operator is testing the network to ensure that subscribers receive the best possible quality, and thereby has some value. In either case, it is necessary to ensure that the symbol that is degraded does not contain customer bits or that the probability of causing an uncorrectable error is acceptably low.

A cable television network may be used to provide bidirectional voice communication, similarly to the public telephone network. In this case, the subscriber's telephone instrument is not connected to the public telephone network but is connected through a suitable adapter to the television cable network. The adapter digitizes the subscriber's outgoing voice message and employs it to modulate a carrier, and similarly detects and converts to analog form an incoming digitized voice message. The headend is connected to the telephone instrument of the other party to the call through another network, which might be the public telephone network or include another cable distribution system. In either case, voice messages are transmitted bidirectionally between the subscriber node and the headend over the cable network by digitizing the voice messages and modulating a carrier with the digitized voice messages. The test method described herein can be used for testing a transmission channel used for voice transmission by providing a diagnostic function in the equipment at the subscriber node. The diagnostic function may be added to the functions performed in the subscriber's telephone/cable adapter or may alternatively be provided by a separate diagnostic receiver.

Bidirectional voice transmissions tend to be bursty, but excessive latency in response may be objectionable to the user. Accordingly, test packets should only be used during transmission if they are short enough that they will not cause excessive latency. Alternatively, since voice transmissions tend to be relatively short and have a protocol for starting and finishing each transmission session, test packets may be sent when setting up a call, tearing down a call, or during idle times.

FIG. 5 illustrates schematically a public telephone network including a node 110, such as a central office or fiber node, and subscriber lines 114 extending from the central node 110 to respective subscriber nodes 118. Analog voice traffic may be carried on the lines 114. Digital data may also be transmitted over the subscriber lines. For example, the central node may be connected to an internet service provider and provide for data transmission between a subscriber node and the ISP. In accordance with an xDSL protocol, such as ADSL (asynchronous digital subscriber line), the digital data is used to modulate one or more carriers, each having a frequency outside the audio range and the digital data can then be transmitted concurrently with the analog voice traffic. In this case, the central node includes an xDSL transceiver and the subscriber node also includes an xDSL transceiver, for transmitting data between the central node and the subscriber node using the ADSL protocol.

The invention may be used to test the subscriber lines 114 to ensure that the digital data can be transmitted error free. At the subscriber node, the xDSL transceiver includes, or is provided with, a diagnostic receiver which operates similarly to the diagnostic cable receiver illustrated in FIG. 3. This provides a technique for detecting impairments in the transmission channel from the central node to individual subscriber nodes before the transmission channel is degraded to such an extent that error protected data packets cannot be recovered at the subscriber node. The other functions described with reference to FIGS. 3 and 4, such as transmission of messages to a central location and remote classification of impairments, apply to the system described with reference to FIG. 5.

In the case of data transmission, it is much simpler to include impaired packets in the transmission because data transmissions are usually bursty. By using a broadcast protocol, many subscriber lines can be tested in parallel by sending test packets to all subscriber nodes simultaneously. When testing an individual subscriber line, it is necessary to control operation of the impairments generator to ensure that the packet address will not be impaired, so that the subscriber node can correctly identify a packet intended for it. Rather, only the data inside the packet should be impaired.

It will be appreciated that the invention is not restricted to the particular embodiments that have been described, and that variations may be made therein without departing from the scope of the invention as defined in the appended claims and equivalents thereof. For example, although the description of FIGS. 3 and 4 refers to the return path from the subscriber node 50 to the headend 48 as being the cable that is used for transmission from the headend to the subscriber node, it may instead be implemented by another medium, such as the public switched telephone network.

Claims

1. A method of operating a digital data distribution network having a transmitter and a receiver, wherein digital data is transmitted in error protected packets from the transmitter to the receiver over a transmission path by employing the digital data to modulate at least one carrier and impressing the modulated carrier on the network, said method comprising:

(a) generating an error protected data packet for transmission over the transmission path,

(b) impairing the transmission path to a selected extent upstream of a transmission path segment that is to be tested,

(c) transmitting the data packet over the transmission path,

(d) receiving the data packet at the receiver, and

(e) determining whether the received data packet is error free.

2. A method according to claim 1, further comprising, if the transmission path is not error free, transmitting a message from the receiver to the transmitter.

3. A method according to claim 1, wherein step (c) comprises progressively increasing the extent to which the transmission path is impaired.

4. A method according to claim 1, wherein the network has a plurality of receivers and digital data is transmitted to the receivers over respective transmission paths, step (b) comprises transmitting the data packet to the receivers over the respective transmission paths, step (d) comprises receiving the data packet at each receiver, and step (e) comprises determining at each receiver whether the received data packet is error free.

5. A method according to claim 4, further comprising, if the transmission path to a selected receiver is not error

free, transmitting a message from the selected receiver to the transmitter, and analyzing messages received at the transmitter.

6. A method according to claim 4, wherein the method comprises detecting receivers that report higher than average error rates and comparing the transmission paths to the respective receivers in a manner such as to derive information from the receivers that report higher than average error rates.

7. A method according to claim 4, wherein step (c) comprises progressively increasing the extent to which the transmission path is impaired and the method further comprises:

transmitting a message from a receiver to the transmitter if the transmission path to that receiver is not error free, and

correlating messages received at the transmitter with the extent to which the transmission path is impaired.

8. A method of operating a digital data distribution network having a transmitter and a receiver, wherein digital data is transmitted in error protected packets from the transmitter to the receiver over a transmission path by employing the digital data to modulate at least one carrier and impressing the modulated carrier on the network, said method comprising:

(a) generating an error protected data packet for transmission over the transmission path,

(b) transmitting the data packet over the transmission path as an analog signal,

(c) receiving the analog signal at the receiver,

(d) recording the analog signal received at the receiver, and

(e) transmitting the record of the analog signal to a remote location for analysis.

9. A method according to claim 8, wherein step (e) comprises transmitting the record of the analog signal to the transmitter for analysis.

10. A method according to claim 8, wherein step (d) comprises digitizing the analog signal and the method further comprises deriving digital data from the digitized signal, forming a data packet from the digital data derived from the digitized signal, transmitting the data packet to the transmitter as an analog signal, digitizing the analog signal received at the transmitter, and processing the digitized signal at the transmitter.

11. A method according to claim 10, wherein the step of processing the digitized signal at the transmitter comprises deriving the error vector waveform for the transmission path to the receiver.

12. A method according to claim 8, wherein the network has a plurality of receivers and digital data is transmitted to the receivers over respective transmission paths, step (d) comprises digitizing the analog signal at each receiver, and the method further comprises deriving digital data from the digitized signal at each receiver, forming data packets at the respective receivers from the digital data derived from the digitized signal at each receiver, transmitting the data packets to the transmitter as analog signals, digitizing the analog signals received at the transmitter, and processing the digitized signals at the transmitter, and the step of processing the digitized signal at the transmitter comprises deriving the error vector waveforms for the transmission paths to at least first and second receivers and deriving the correlated error vector function for the transmission paths to the first and second receivers.

13. A method according to claim 8, wherein the network has a plurality of receivers and digital data is transmitted

to the receivers over respective transmission paths, and the method comprises deriving digital data from the digitized signal at each receiver, transmitting the digital data to the transmitter, and analyzing data received from multiple receivers to extract the locations of uncorrelated impairments.

14. A method according to claim 8, comprising processing the digitized signal to classify an impairment in the transmission path and testing the transmission margin of the transmission path with respect to the impairment by impairing the transmission path using that impairment.

15. A method according to claim 8, further comprising impairing the transmission path to a selected extent upstream of a transmission path segment to be tested.

16. A method of operating a digital data distribution network having a transmitter and a receiver, wherein digital data is transmitted in error protected packets from the transmitter to the receiver over a transmission path by employing the digital data to modulate at least one carrier and impressing the modulated carrier on the network, said method comprising:

generating an error protected data packet for transmission over the transmission path,

impairing the transmission path to a selected extent upstream of a transmission path segment that is to be tested,

transmitting the data packet over the transmission path,

receiving the data packet at the receiver, and

counting bit errors in the received data packet.

17. A method of operating a digital data distribution network having a transmitter and a receiver, wherein digital data is transmitted in error protected packets from the transmitter to the receiver over a transmission path by employing the digital data to modulate a carrier and

impressing the modulated carrier on the network, said method comprising:

(a) generating an error protected data packet for transmission over the transmission path,

(b) impairing the transmission path to a selected extent upstream of a transmission path segment that is to be tested,

(c) transmitting the data packet over the transmission path as an analog signal, and

(d) receiving the analog signal at the receiver.

18. A method according to claim 17, wherein the step of impairing the transmission path comprises impairing the transmission path by reducing its signal-to-noise ratio.

19. A method according to claim 17, wherein the step of impairing the transmission path comprises impairing the transmission path by reducing its frequency response.

20. A method according to claim 17, wherein the step of impairing the transmission path comprises impairing the transmission path by reducing its phase response.

21. A method according to claim 17, wherein the step of impairing the transmission path comprises impairing the transmission path by reducing its impulse response.

22. A method according to claim 17, wherein the step of impairing the transmission path comprises impairing the transmission path by introducing phase noise or jitter.

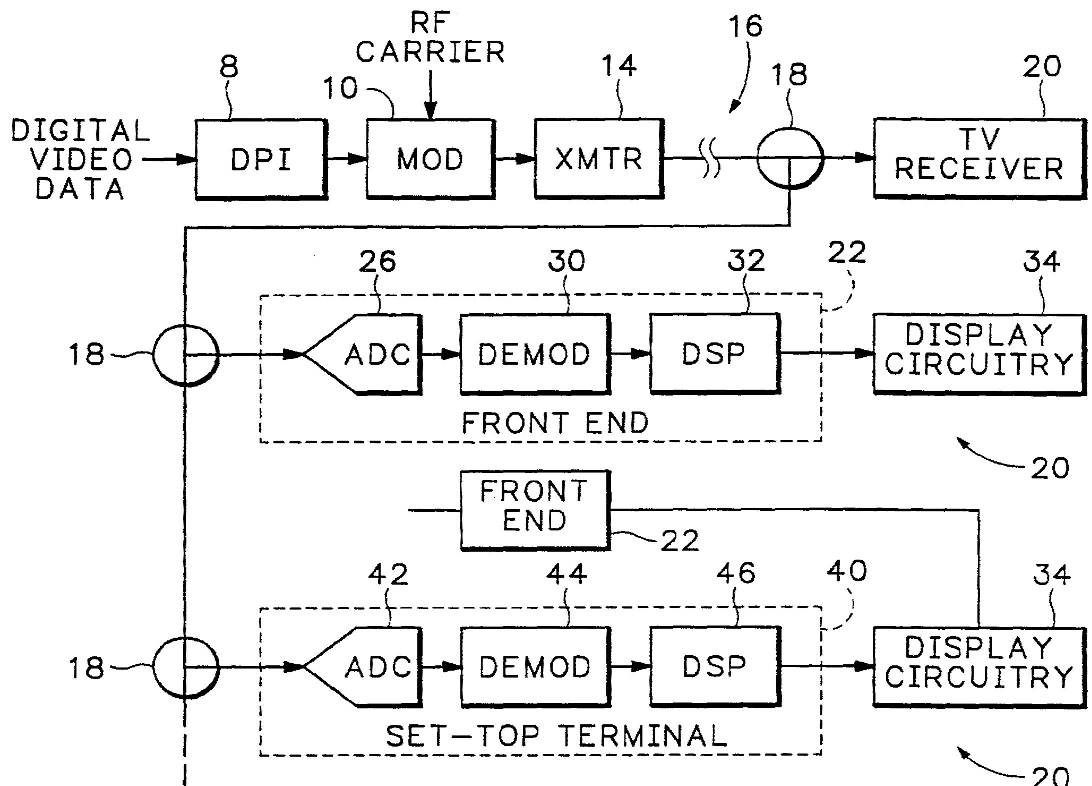
DIGITAL VIDEO DATA
8
DPI
10
RF CARRIER
MOD
14
XMTR
16
18
20
TV RECEIVER
26
ADC
30
DEMOD
32
DSP
22
34
DISPLAY CIRCUITRY
18
FRONT END
20
FRONT END
22
42
ADC
44
DEMOD
46
DSP
40
34
DISPLAY CIRCUITRY
18
SET-TOP TERMINAL
20

FIG.1

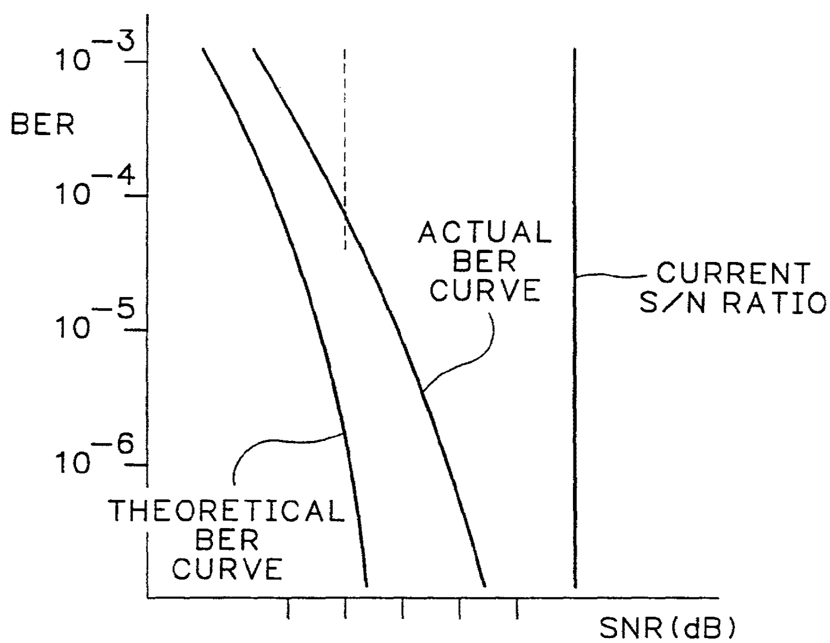
10^{-3}
BER
10^{-4}
10^{-5}
10^{-6}
ACTUAL BER CURVE
CURRENT S/N RATIO
THEORETICAL BER CURVE
SNR(dB)

FIG.2

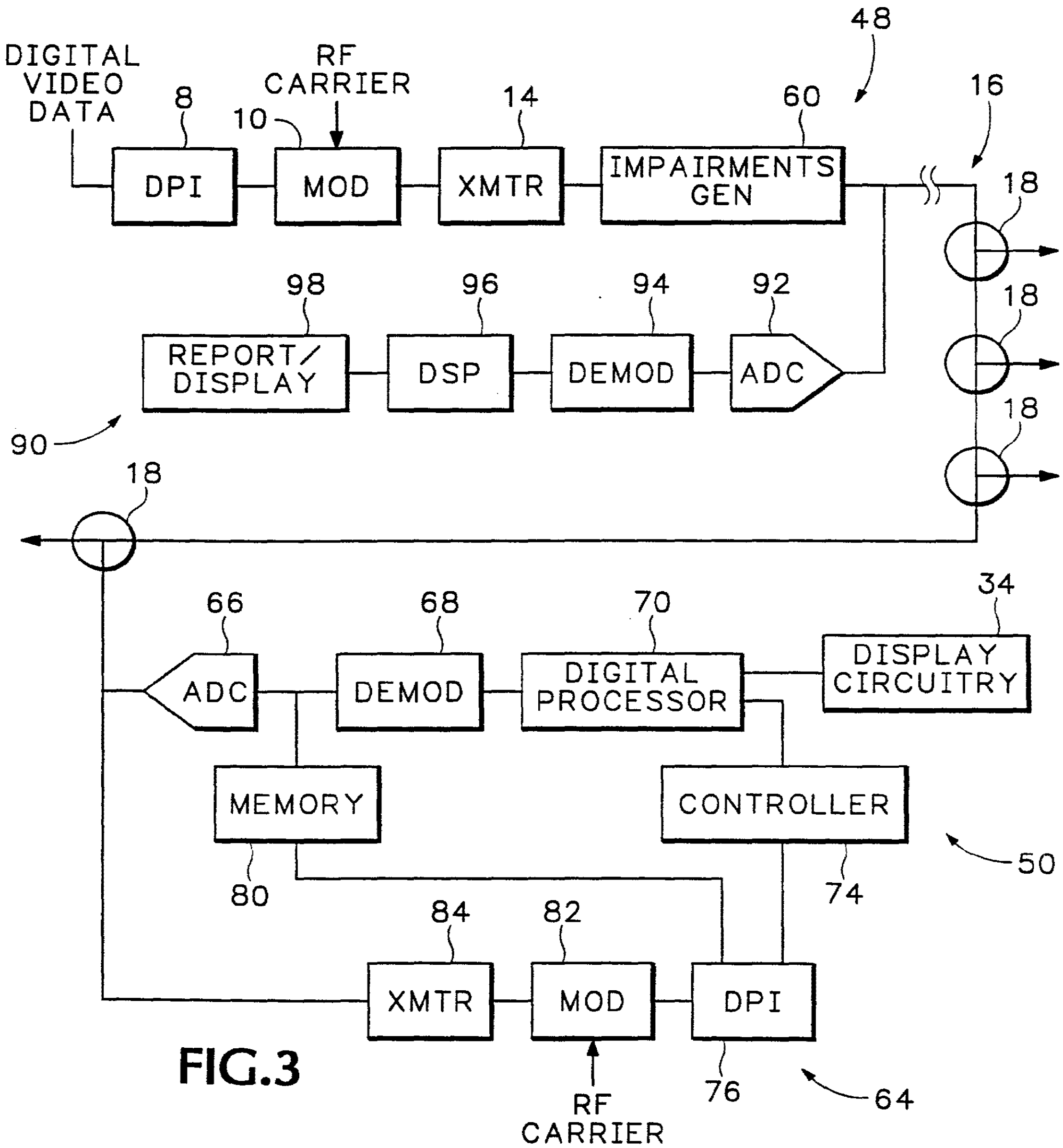
DIGITAL VIDEO DATA
RF CARRIER
48
8
10
14
60
16
DPI
MOD
XMTR
IMPAIRMENTS GEN
18
98
96
94
92
18
REPORT/ DISPLAY
DSP
DEMOD
ADC
90
18
18
66
68
70
34
ADC
DEMOD
DIGITAL PROCESSOR
DISPLAY CIRCUITRY
MEMORY
CONTROLLER
50
80
74
84
82
XMTR
MOD
DPI
76
64
RF CARRIER

FIG.3

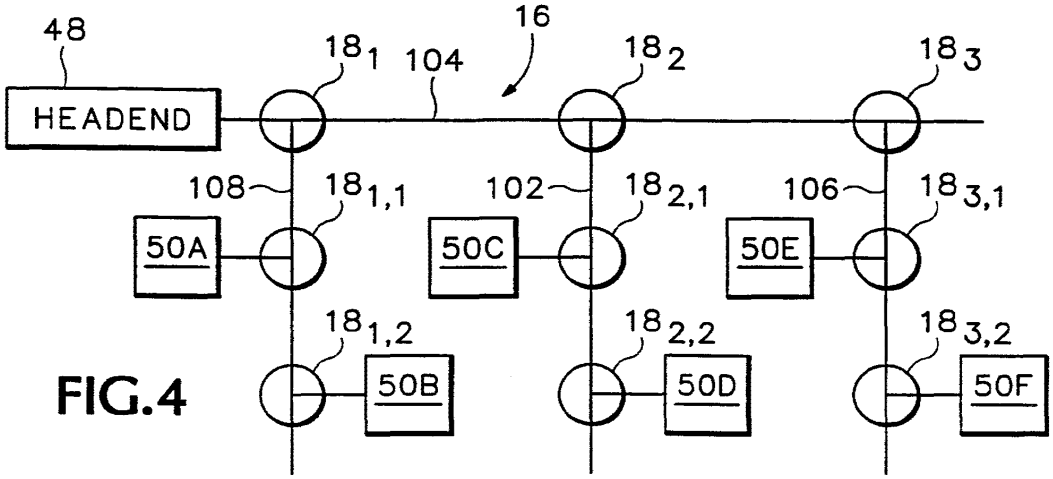
48
HEADEND
16
18 1
104
18 2
18 3
108
18 1,1
102
18 2,1
106
18 3,1
50A
50C
50E
18 1,2
18 2,2
18 3,2
50B
50D
50F

FIG.4

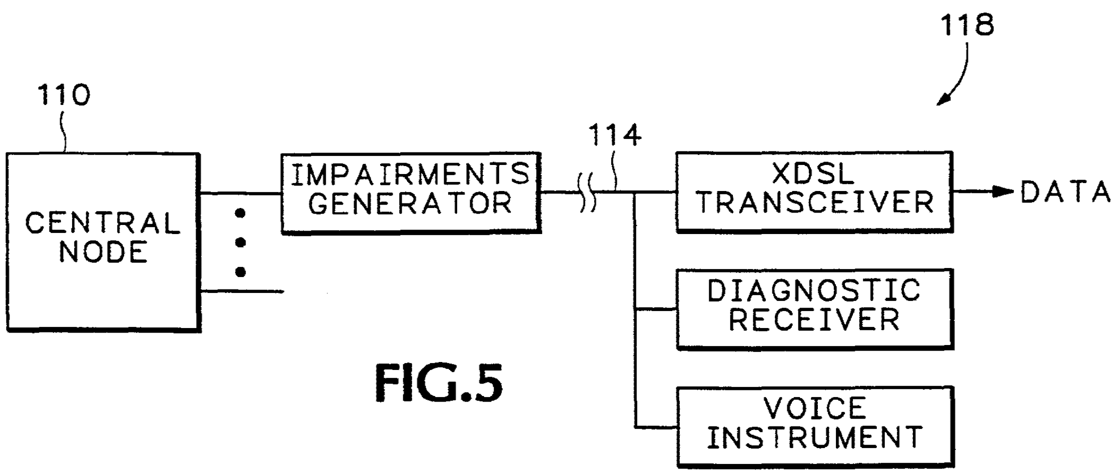
118
110
CENTRAL NODE
IMPAIRMENTS GENERATOR
114
XDSL TRANSCEIVER
DATA
DIAGNOSTIC RECEIVER
VOICE INSTRUMENT

FIG.5

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INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 : G06F 9/00, H04B 3/23, 3/32	A1	(11) International Publication Number: WO 99/63427 (43) International Publication Date: 9 December 1999 (09.12.99)

(21) International Application Number: PCT/US99/11052

(22) International Filing Date: 19 May 1999 (19.05.99)

(30) Priority Data:
09/086,386 29 May 1998 (29.05.98) US

(71) Applicant: GTE LABORATORIES INCORPORATED [US/US]; 1209 Orange Street, Wilmington, DE 19801 (US).

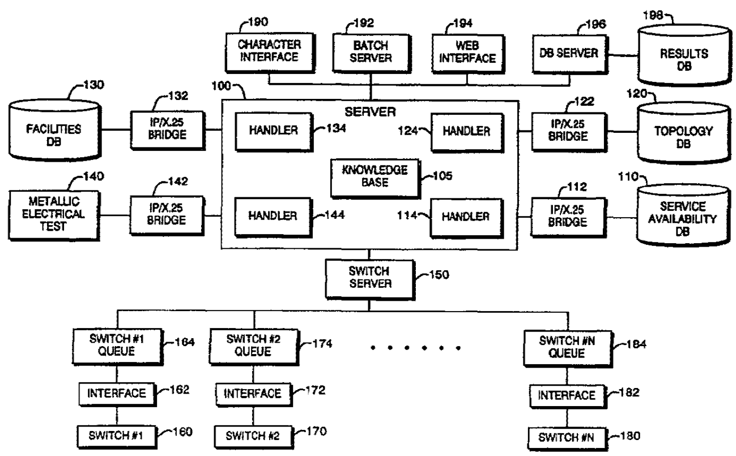
(72) Inventors: EICHEN, Elliot; 41 Mary Street, Arlington, MA 02174 (US). BROOKS, David, L.; 11 Vose Hill Road, Maynard, MA 01754 (US).

(74) Agents: SUCHYTA, Leonard, Charles et al.; GTE Service Corporation, 600 Hidden Ridge, MC HQE03G13, Irving, TX 75038 (US).

(81) Designated States: AE, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, CA, CH, CN, CU, CZ, DE, DK, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, UA, UG, UZ, VN, YU, ZA, ZW, ARIPO patent (GH, GM, KE, LS, MW, SD, SL, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published
With international search report.

(54) Title: METHOD AND APPARATUS FOR DIGITAL SUBSCRIBER LOOP QUALIFICATION



(57) Abstract

A system and methodology for qualifying a twisted pair copper loop for digital subscriber loop services are described. The system automatically queries telecommunications provider database records and/or requests measurements from network switching equipment or testing systems to obtain data regarding the twisted pair copper loop, such as loop length, electrical characteristics, and other loop topology characteristics such as wire gauge, the presence of load coils (88, 90), and the presence of bridge taps (84, 86). The system determines which digital subscriber loop services are available for the copper loop based on the combination of all data obtained. The system may be implemented in part as an expert system with a knowledge base of qualification rules used in the decision–making process.

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METHOD AND APPARATUS FOR DIGITAL SUBSCRIBER LOOP QUALIFICATION

Technical Field

The present invention relates to digital subscriber loop technology and, more specifically, to the qualification of existing twisted pair copper loops for digital subscriber loop service.

Background Art

Digital subscriber loop technology is the digital encoding of all information transmitted on the local loop, *i.e.*, the connection between a customer's premises (home, office, etc.) and a telecommunications provider's central office serving the customer's premises. Most existing local loops in the United States and throughout the world are twisted pair copper loops, originally designed for analog service, or plain old telephone service (POTS). With digital subscriber loop technology, high speed access to the Internet, advanced telephony functions, and multimedia services is possible over the twisted pair copper access network. Digital subscriber systems can provide data from speeds of 64 kb/second in both upstream and downstream directions to over 10 Mb/second in a single direction. Digital subscriber loop technology, often referred to as "xDSL" where x stands for any of a number of letters, includes the following:

ADSL, Asymmetric Digital Subscriber Loop
VDSL, Very High-Speed Digital Subscriber Loop
HDSL, High Data Rate Digital Subscriber Loop
SDSL, Symmetric Digital Subscriber Loop
IDSL, ISDN-based Digital Subscriber Loop
RADSL, Rate Adaptive Digital Subscriber Loop
ISDN, Integrated Digital Service Network

Some of these digital subscriber loop technologies (*e.g.*, HDSL, ISDN, and in particular ADSL) have been standardized by various standards bodies with respect to modulation format, bandwidth, and embedded operations channels, while others have not been standardized and are available from different vendors in a wide variety of modulation formats, upstream/downstream bandwidths, and operation channels.

As illustrated in Figure 1, digital subscriber loop technology consists of two terminal endpoints (TEs) 10 and 20, which provide conversion, modulation, transmission, and reception of data, and copper loop 30 connecting TEs 10 and 20. TE 10 is typically owned and operated by the service provider, while TE 20 is typically at the customer's premises. In the United States, TE 20 is typically owned or rented by the customer, while in most other parts of the world TE 20 is typically owned and operated by the service provider. In addition, the digital subscriber loop topology can include terminal equipment, such as a repeater, between the two terminal endpoints to provide additional network flexibility or to boost signal strength and transmission distances. For example, Figure 2 illustrates network terminal 70 in copper loop 60 between TEs 40 and 50.

Digital subscriber loop services, however, cannot be carried over all twisted pair copper loops that support POTS service. The various digital subscriber loop technologies have complex (real and imaginary) signal attenuation restrictions that depend upon downstream (to the customer) and upstream (from the customer) bandwidth, modulation format, and receiver sensitivity for a particular chip set used by a vendor terminal endpoint equipment. Signal attenuation itself depends on several factors, including the length and gauge of the copper wires contained in the loop, the environment in which the copper wires are placed (including temperature variations), and the quality of connections (*e.g.*, splices and terminal connections) that attach the different sections of wire contained in a given loop. Digital subscriber loop technologies also have restrictions on loop topology, such as the position and number of bridge taps and load coils, and restrictions on services provided in adjacent copper pairs in the same binder group (*i.e.*, a group of twisted pairs bundled together) because of crosstalk between pairs and overlapping frequency spectrums.

Figure 3 illustrates a typical copper loop between central office (CO) 80 and terminal endpoint 82, made up of several different lengths of wire of different gauges spliced together. One leg of the loop terminates at terminal endpoint 82, while two other legs are unterminated, resulting in bridge taps 84 and 86. The loop in Figure 3 also includes two load coils, 88 and 90, as well as cross connect 92.

As an example of loop topology requirements, a loop is restricted to less than approximately 5.25 km of 24 gauge wire when the digital service is provided at the rate of 1.5 Mb/second downstream and 80 kb/second upstream for a commonly available chip sent that uses carrierless amplitude phase (CAP) modulation for ADSL. For this modulation format and bandwidth allocation, if there is an analog carrier POTS service in the same wire binder group, the ADSL modulation will interfere with the analog carrier, effectively destroying the POTS service. Similarly, if there is a T1 carrier system in the same wire binder group, the T1 service will interfere with the ADSL modulation, nullifying the digital subscriber service, but typically not affecting the T1 service. The number of copper pairs and the potential for crosstalk in a binder group depends on the type and manufacturer of the copper cable.

Today, when a customer wishes to order a digital subscriber loop service, the local telecommunications service provider must determine whether the customer's existing twisted pair copper loop can support the requested digital subscriber loop service at the desired bandwidth. This can be a difficult and time-consuming task to perform manually because of the many restrictions on loop topology and services just described. All necessary data may not be available to a person trying to qualify a loop for digital subscriber loop services, particularly because telecommunications providers often have data in many different databases or stored in paper records. Even if data is available, data concerning outside plant information such as loop length and topology is often out of date. Also, certain metallic loop electrical data is not stored in a database and can only be determined by a measurement or test system.

It is desirable, therefore, to provide a system and methodology for determining which digital subscriber loop technologies can be supported by a particular twisted pair copper loop. It is more desirable to qualify a copper loop for digital subscriber loop services on the basis of real-time electrical measurements as well as records stored in telecommunications provider databases. It is even more desirable to provide an automated system for digital subscriber loop qualification that economically determines which digital subscriber loop technologies can be supported by a copper loop. It is also desirable to implement such a system as an expert system containing a knowledge base of rules.

Disclosure of Invention

The present invention satisfies those desires by providing a system and methodology for qualifying a twisted pair copper loop for digital subscriber loop services. The system automatically queries telecommunications provider database records and/or requests measurements from network switching equipment or testing systems to obtain information regarding the twisted pair copper loop in question. The system then determines which digital subscriber loop services are available for the copper loop based on the combination of all information obtained.

A method consistent with the present invention for qualifying a twisted loop pair for a digital subscriber service comprises the steps of receiving as input a unique identifier corresponding to the loop, determining a topology corresponding to the loop, and determining whether the loop meets topology restrictions of the digital subscriber service. Another method consistent with the present invention comprises the steps of receiving data corresponding to physical characteristics of the loop and applying a plurality of rules to the data to determine whether the loop is suitable for the digital subscriber service. Other methods consistent determine whether electrical characteristics of the loop meet restrictions of the digital subscriber service and whether services provided on other cable pairs in the same binder group with the loop are compatible with the digital subscriber service.

Systems are also provided for carrying out the methodologies of the present invention.

The advantages accruing to the present invention are numerous. A loop qualification system and method consistent with the present invention reduce the time for determining which digital subscriber loop services a particular copper loop supports from several hours to a few minutes. A system and method consistent with the present invention also provide a substantially more accurate result, in part because they use real-time electrical measurements to determine many topological characteristics of the copper loop.

The above desires, and other desires, features, and advantages of the present invention will be readily appreciated by one of ordinary skill in the art from the following detailed description of the preferred implementations when taken in connection with the accompanying drawings.

Brief Description of Drawings

Figure 1 illustrates digital subscriber loop technology connecting two terminal endpoints;

Figure 2 illustrates digital subscriber loop technology with network terminal equipment between two terminal endpoints;

Figure 3 illustrates a typical digital subscriber loop topology;

Figure 4 illustrates the architecture of a loop qualification system consistent with the present invention; and

Figure 5 is a flow chart of a method for qualifying loops consistent with the present invention.

Best Mode for Carrying Out the Invention

A system consistent with the present invention automatically qualifies twisted pair copper loops for digital subscriber loop services. Generally, a method for qualifying loops for digital subscriber loop services consistent with the present invention includes at least four types of qualification:

(1) Service Availability: Is the point at which the copper loop terminates equipped to provide the requested digital subscriber service?

(2) Length Qualification: Which digital subscriber loop services at which bandwidths can be provided given the length of the loop?

(3) Line Qualification: Is the loop physically suitable for use by a digital subscriber loop technology? Is the service currently provisioned on the loop compatible with digital subscriber loop service?

(4) Are the services provided on the other twisted pairs in the same binder group with the loop spectrally compatible with digital subscriber loop services?

In order to answer these loop qualification questions, a system consistent with the present invention combines results obtained from testing the copper loop, results from queries of telecommunications provider database records, and information regarding the transmission and receiver characteristics of the digital subscriber.

Figure 4 illustrates the architecture of a system for qualifying loops consistent with the present invention, which may be implemented, for example, as an expert system using a conventional client-server architecture known in the art. The expert system is implemented in software residing on server 100 and performs loop qualification by combining input from a number of information sources with rules contained in knowledge base 105. Specifically, server 100 obtains information from service availability database 110, topology database 120, facilities database 130, and metallic electrical test system 140. Databases 110, 120, and 130, and test system 140 are typically owned and operated by the local telecommunications provider. It will be recognized by one skilled in the art that each database shown in Figure 4 may actually consist of several smaller databases or, alternatively, that databases may be combined, since each telecommunications provider organizes its data into databases in different ways. Server 100 interfaces to the databases and test system via a suitable communications protocol such as IP or X.25, provided by interfaces 112, 122, 132, and 142. Server 100 additionally includes software handler modules for receiving and processing information obtained from databases 110, 120, and 130, and test system 140.

Server 100 also receives information and test results directly from central office switches in the local network, three of which are shown in Figure 4 as switches 160, 170, and 180 for illustrative purposes. Server 100 is coupled to switch server 150, which is coupled to switch queues 164, 174, and 184, corresponding to switches 160, 170, and 180, respectively. Switch queues 164, 174, and 184 access data from switches 160, 170, and 180 via interfaces 162, 172, and 182, respectively. It will be recognized by one skilled in the art that switch server 150 need not be separate from server 100.

Consistent with the present invention, a user may access server 100 through either the graphical user interface of client 194, *e.g.*, a World Wide Web-based client, or character interface 190, *e.g.*, a VT100 character interface. Regardless of the interface used, a user will typically enter a unique number (*e.g.*, a telephone directory number (TDN) or an IP address) or identifier (*e.g.*, a circuit identifier) associated with the copper loop for which qualification is desired. A system consistent with the present invention also includes batch server 192, which allows qualification of numerous loops to be performed in batch, and database server 196 for storing results in results database 198.

Figure 5 is a flow chart illustrating a method for qualifying loops for digital subscriber loop services consistent with the present invention. Consistent with an embodiment of the present invention, the method is performed by software residing on server 100. The process begins by receiving as input a unique identifier corresponding to the copper loop to be qualified for digital subscriber services (step 200). The unique identifier may be a telephone directory number (TDN) as shown in Figure 5, or any other unique identifier such as an IP address or a circuit identifier. Also, server 100 may receive the identifier from any input source, including character interface 190 or web interface 194 (if a human user is accessing the system through an interface) and batch server 192 (if several qualification requests have been entered for batch processing). Most of the remaining steps in the process use the unique loop identifier to retrieve information regarding the loop.

Once receiving a loop identifier, the qualification process continues by determining whether digital subscriber loop services are available for the loop (step 210). Consistent with the present invention, the server makes this determination by querying service availability database 110 to determine whether the local telecommunications provider provides xDSL services from the office serving the customer's location. If xDSL service is not available, loop qualification terminates. If xDSL service is available, processing continues to step 220. In an alternate method consistent with the present invention, the server may choose to continue the loop qualification process although xDSL service is not available.

Next, the process determines whether the loop is on a working pair (step 220) by querying facilities database 130. Some measurement tests performed by a loop qualification method consistent with the present invention require that the loop be on a working pair. If the loop is not on a working pair, the server either terminates loop qualification (as shown in Figure 5) or chooses to continue loop qualification, although not all tests will be available for the loop. Alternatively, the loop may be temporarily assigned to line equipment and a test number so that loop qualification may be performed.

If loop qualification continues, the server determines whether the current service on the loop is compatible with xDSL service (step 230). For example, in the United States the current service cannot be T1 or ISDN. Consistent with the present invention, the server performs this step by querying facilities database 130. As discussed above, it should be apparent to one skilled in the art that, although the queries in steps 220 and 230 both access databases with information regarding facilities, the facilities database shown in Figure 4 (database 130) may consist of several smaller databases, so that the queries of steps 220 and 230 access two different, smaller databases. If the current service is not compatible, loop qualification ends. If the current service is compatible, then flow proceeds to several data collection steps. In an alternate method consistent with the present invention, the server may choose to continue the loop qualification process although the current service is not compatible with xDSL service.

A method consistent with the present invention performs some or all of data collection steps 240, 250, 260, and 270. These steps are not necessarily performed in a particular order, and some steps may be performed simultaneously. For example, Figure 5 shows steps 240 and 250 being performed at the same time as steps 260 and 270. Each of these steps involves obtaining information about the loop to be qualified from a database or a test or measurement system in the network, and all of the information obtained is used as input to step 280, which applies a plurality of rules to the information to model the response of the network and determine which digital subscriber services are available on the loop.

In step 240, the server queries topology database 120 using the unique loop identifier (*e.g.*, TDN or IP address) to obtain a variety of loop topology data. In particular, the server requests length and gauge of wire on the loop for each loop segment, cable type, the location of load coils on the loop, and the location and length of bridge taps on the loop. For example, the loop topology shown in Figure 3 is an example of data that may be obtained from topology database 120. As described above, topology database 120 may consist of several smaller databases, each of which contains different information. Step 240 may also include a query of a separate database (not shown in Figure 4) that stores recent measurements of the loop length. This data may be more accurate than a topology database operated by the telecommunications provider for storing many different types of loop topology data.

Referring again to Figure 5, in step 250 the server queries facilities database 130 using the unique loop identifier to determine the services on other cable pairs in the same binder group as the loop to be qualified. This information will be used in step 280 to determine whether xDSL services are spectrally compatible with the services on the other cable pairs in the binder so that crosstalk will not degrade service quality.

In step 260, the server requests measurements from metallic electrical test system 140, which is a remote test system such as 4TEL, manufactured by Teradyne, Inc., or Mechanized Loop Test (MLT), manufactured by Lucent Technologies. Consistent with the present invention, the server requests a measure of loop length and/or loop capacitance, which can be converted to loop length using a known mathematical relationship. The server also requests measures of longitudinal balance and wideband and narrowband electrical ingress which will be used in step 280 to determine the suitability of the loop for digital subscriber loop services. As described earlier, tests in step 260 may not be performed if the loop is not on a working pair.

In step 270, the server requests a load coil detection measurement to determine if there are any load coils in the loop. This measurement can be performed at the end office switch at which the loop terminates (*e.g.*, switch 160, 170, or 180 in Figure 4) or by metallic electrical test system 140. If the server obtains the measurement from the switch, switch server 150 receives measurements from queues 164, 174, and 184, and controls server 100's access to switch measurements. Examples of load coil detection measurements known in the art are a swept frequency measurement and a time domain reflectometry measurement. As descirbed earlier, tests in step 270 may not be performed if the loop is not on a working pair.

All of the information obtained in steps 240, 250, 260, and 270 from database queries and test and measurement systems is input to step 280. Consistent with the present invention, in step 280 an expert system resident on server 100 combines the results of steps 240, 250, 260, and 270 with a plurality of qualification rules from knowledge base 105 and information on network equipment stored in a database (not shown for the sake of clarity) to model the response of the network for the various digital subscriber loop services available to the subscriber. The expert system also determines, for each of the available digital subscriber loop services (*e.g.*, ADSL, VDSL, etc.), how much bandwidth can be supported in both upstream and downstream directions.

Consistent with the present invention, the qualification rules in knowledge base 105 are not limited to any particular set. The rules may range from the simple (*e.g.*, a loop with one or more load coils does not qualify for a digital subscriber loop service) to the more complex (*e.g.*, for a certain type of terminal equipment and a particular digital subscriber loop service with given upstream and downstream bandwidth, a combination of wire length and gauge limits can be calculated according to mathematical relationships to satisfy given signal attenuation and/or bit error rate requirements).

Consistent with the present invention, there may be a conflict between data retrieved from a database and data measured in real-time using a measurement system or test system. In such cases, knowledge base 105 can also include rules for reconciling the differences. For example, if data retrieved from a database is known not to have been updated recently, then a qualification method consistent with the present invention would rely on measured data, which may be more accurate.

The ultimate output of a system consistent with the present invention is a list of digital subscriber loop service packages that the loop can support. For a particular type of xDSL service (*e.g.*, ADSL), there may be multiple packages, each of which defines a different class of service, including upstream and downstream bandwidth. For example, a loop may be able to support an ADSL package with downstream/upstream bit rates of 640k/272k, but the same loop may not support ADSL with bit rates of 640k/680k because of the loop length and topology. Alternatively, a system consistent with the present invention may determine whether a loop can support a specified digital subscriber loop service with given upstream and downstream bandwidths. In this case, the system user may enter the service type and bandwidth desired. In addition to simply listing qualified services, the system may provide the user with diagnostic information explaining why a particular decision was reached.

It will be apparent to those skilled in this art that various modifications and variations can be made to the loop qualification scheme of the present invention without departing from the spirit and scope of the invention. Other embodiments of the invention will be apparent to those skilled in this art from consideration of the specification and practice of the invention disclosed herein. In particular, the method is not limited to implementation in a client/server architecture or as an expert system. Nor is the invention limited to the user interfaces described. For example, a machine application program interface can provide access to the system from another system or as part of a larger provisioning system. A method consistent with the present invention can also be used to qualify loops for other services whose qualification requires access to database and/or real-time measurements. It is intended that the specification and examples be considered exemplary only, with the true scope and spirit of the invention being indicated by the following claims.

Claims

1. A method for qualifying a twisted pair loop for a digital subscriber service having loop topology restrictions, the method comprising the steps of:

receiving a unique identifier corresponding to the loop;

identifying a topology corresponding to the identified loop; and

determining whether the identified loop meets the loop topology restrictions of the digital subscriber service based on the identified topology.

2. The method of claim 1 further including the step of computing a bandwidth that can be supported on the loop in response to the topology.

3. The method of claim 1 wherein the identifying step includes the substeps of determining a length corresponding to the loop and determining a gauge corresponding to the loop.

4. The method of claim 3 wherein the substep of determining a length includes the substeps of requesting a capacitance measurement of the loop from a metallic electrical testing system and converting the capacitance measurement into the length.

5. The method of claim 3 wherein the substep of determining a length includes the substep of requesting a length measurement from a metallic electrical testing system.

6. The method of claim 3 wherein the substep of determining a length includes the substep of querying a database containing a recent length measurement of the loop.

7. The method of claim 3 wherein the substep of determining a length includes the substep of querying a database containing a known length of the loop.

8. The method of claim 3 wherein the substep of determining a gauge includes the substep of querying a topology database containing the gauge of the loop.

9. The method of claim 1 wherein the loop contains a plurality of loop segments, and wherein the identifying step includes the substeps of determining a length corresponding to each of the plurality of loop and segments and determining a gauge corresponding to each of the plurality of loop segments.

10. The method of claim 9 wherein the substep of determining a length includes the substep of querying a database containing the length of each of the plurality of loop segments.

11. The method of claim 9 wherein the substep of determining a gauge includes the substep of querying a database containing the gauge of each of the plurality of loop segments.

12. The method of claim 1 wherein the identifying step includes the substep of querying a database for the position of a bridge tap in the loop.

13. The method of claim 1 wherein the identifying step includes the substep of querying a database for the length of a bridge tap in the loop.

14. The method of claim 1 wherein the identifying step includes the substep of querying a database for the number of bridge taps in the loop.

15. The method of claim 1 wherein the identifying step includes the substep of querying a database for the position of a load coil in the loop.

16. The method of claim 1 wherein the identifying step includes the substep of querying a database for the number of load coils in the loop.

17. The method of claim 1 wherein the identifying step includes the substeps of requesting a load coil measurement of the loop from a test system at a switch connected to the loop.

18. The method of claim 17 wherein the load coil measurement is a swept frequency measurement.

19. The method of claim 17 wherein the load coil measurement is a time domain reflectometry measurement.

20. A method for qualifying a twisted pair loop for a digital subscriber service, the method comprising the steps of:

receiving a unique identifier corresponding to the loop;

identifying a first cable pair and a binder corresponding to the identified loop having the first cable pair corresponding to the identified loop and a second cable pair; and

determining whether services provided on the second cable pair are compatible with the digital subscriber service.

21. The method of claim 20 wherein the identifying step includes the substep of querying a database correlating the binder to the first and second cable pairs and services provided on the cable pairs, and wherein the determining step includes the substep of querying the database.

22. A method for qualifying a twisted pair loop for a digital subscriber service, the method comprising the steps of:

receiving a unique identifier corresponding to the loop;

identifying a current service on the identified loop; and

determining whether the current service is compatible with the digital subscriber service.

23. The method of claim 22 wherein the identifying step includes the substep of querying a database correlating the identifier to the current service on the loop.

24. The method of claim 22 further comprising the step of determining whether the identifier corresponds to a working loop.

25. A method for qualifying a twisted pair loop for a digital subscriber service having longitudinal balance restrictions, the method comprising the steps of:

receiving a unique identifier corresponding to the loop;

requesting a longitudinal balance measurement of the identified loop from a metallic electrical testing system; and

determining whether the measurement meets the restrictions.

26. A method for qualifying a twisted pair loop for a digital subscriber service having electrical ingress restrictions, the method comprising the steps of:

receiving a unique identifier corresponding to the loop;

requesting an electrical ingress measurement of the identified loop from a metallic electrical testing system; and

determining whether the measurement meets the restrictions.

27. A method for qualifying a twisted pair loop for a digital subscriber service, the method comprising the steps of:

receiving data corresponding to physical characteristics of the loop; and

applying a plurality of qualification rules to the data to determine whether the loop is suitable for the digital subscriber service.

28. The method of claim 27 wherein the receiving step includes the substep of receiving data from a metallic loop electrical test system.

29. The method of claim 27 wherein the receiving step includes the substep of receiving data from a database.

30. The method of claim 27 wherein the receiving step includes the substep of receiving data from a switch connected to the loop.

31. The method of claim 27 wherein the applying step is performed by an expert system.

32. A method for qualifying a twisted pair loop for a telecommunications service, the method comprising the steps of:

receiving data corresponding to physical characteristics of the loop; and

applying a plurality of qualification rules to the data to determine whether the loop is suitable for the telecommunications service.

33. The method of claim 32 wherein the applying step is performed by an expert system.

34. A system for qualifying a twisted pair loop for a digital subscriber service having loop topology restrictions, the system comprising:

an interface for receiving a unique identifier corresponding to the loop;

means for identifying a topology corresponding to the identified loop; and

means for determining whether the identified loop meets the topology restrictions of the digital subscriber service based on the identified topology.

35. The system of claim 34 wherein the identifying means includes means for querying a database.

36. The system of claim 34 wherein the identifying means includes means for requesting a measurement from a test system.

37. A system for qualifying a twisted pair loop for a digital subscriber service comprising:

means for receiving data corresponding to physical characteristics of the loop; and

means for applying a plurality of qualification rules to the data to determine whether the loop is suitable for the digital subscriber service.

38. The system of claim 37 further wherein the means for applying includes an expert system, the expert system including a knowledge base containing the plurality of qualification rules.

39. A system for qualifying a twisted pair loop for a digital subscriber service having loop topology restrictions, said system comprising:

an interface for receiving a unique identifier corresponding to the loop;

a memory comprising a loop qualification program for identifying a topology corresponding to the identified loop, and for determining whether the identified loop meets the loop topology restrictions; and

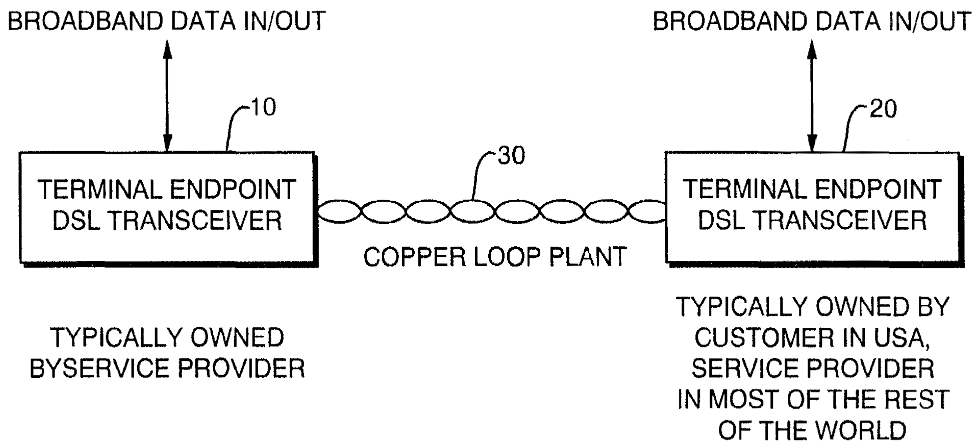
a processor for running the loop qualification program.

40. A system for qualifying a twisted pair loop for a digital subscriber service comprising:

an interface for receiving data corresponding to physical characteristics of the loop;

a memory comprising a knowledge base containing a plurality of rules, and a loop qualification program for applying the plurality of rules; and

a processor for running the loop qualification program.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BROADBAND DATA IN/OUT
10
20
30
TERMINAL ENDPOINT DSL TRANSCEIVER
TERMINAL ENDPOINT DSL TRANSCEIVER
COPPER LOOP PLANT
TYPICALLY OWNED BYSERVICE PROVIDER
TYPICALLY OWNED BY CUSTOMER IN USA, SERVICE PROVIDER IN MOST OF THE REST OF THE WORLD

FIG. 1

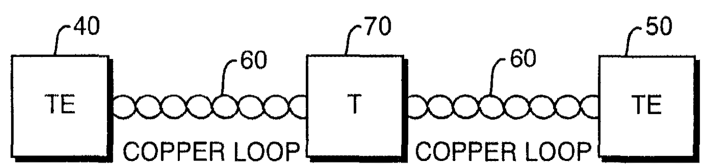
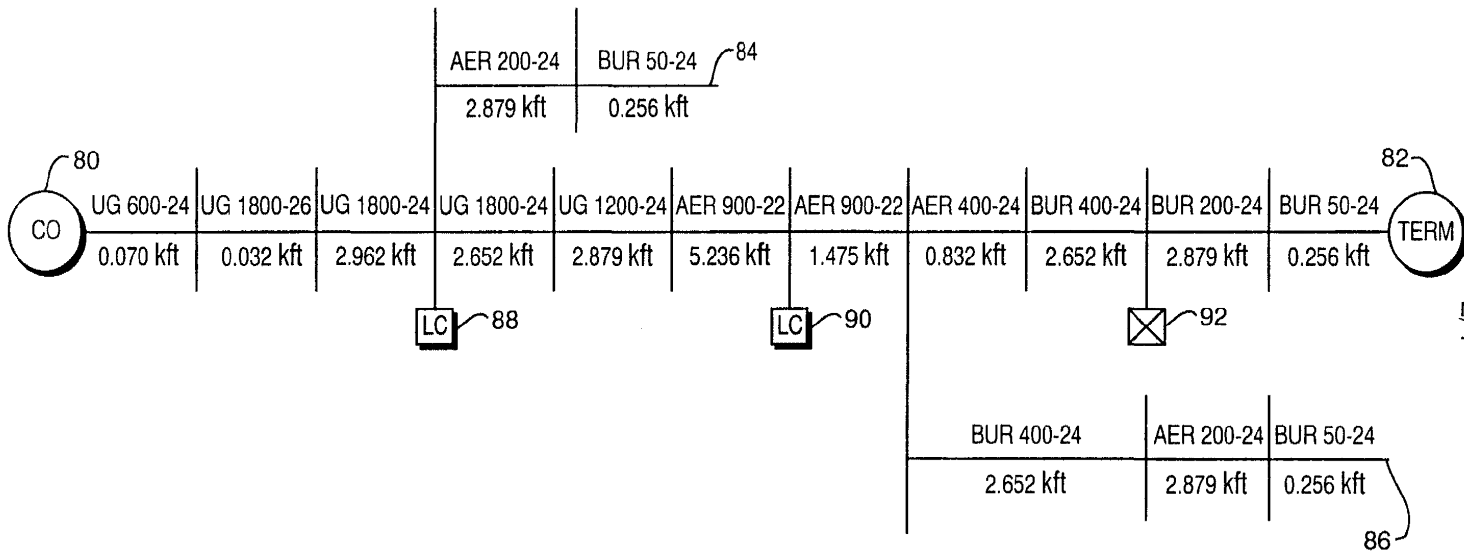

40
70
50
60
60
TE
T
TE
COPPER LOOP
COPPER LOOP

FIG. 2



84
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BUR 50-24
0.256 kft
80
CO
UG 600-24
0.070 kft
UG 1800-26
0.032 kft
UG 1800-24
2.962 kft
UG 1800-24
2.652 kft
UG 1200-24
2.879 kft
AER 900-22
5.236 kft
AER 900-22
1.475 kft
AER 400-24
0.832 kft
BUR 400-24
2.652 kft
BUR 200-24
2.879 kft
BUR 50-24
0.256 kft
82
TERM
LC
88
LC
90
92
BUR 400-24
2.652 kft
AER 200-24
2.879 kft
BUR 50-24
0.256 kft
86

FIG. 3

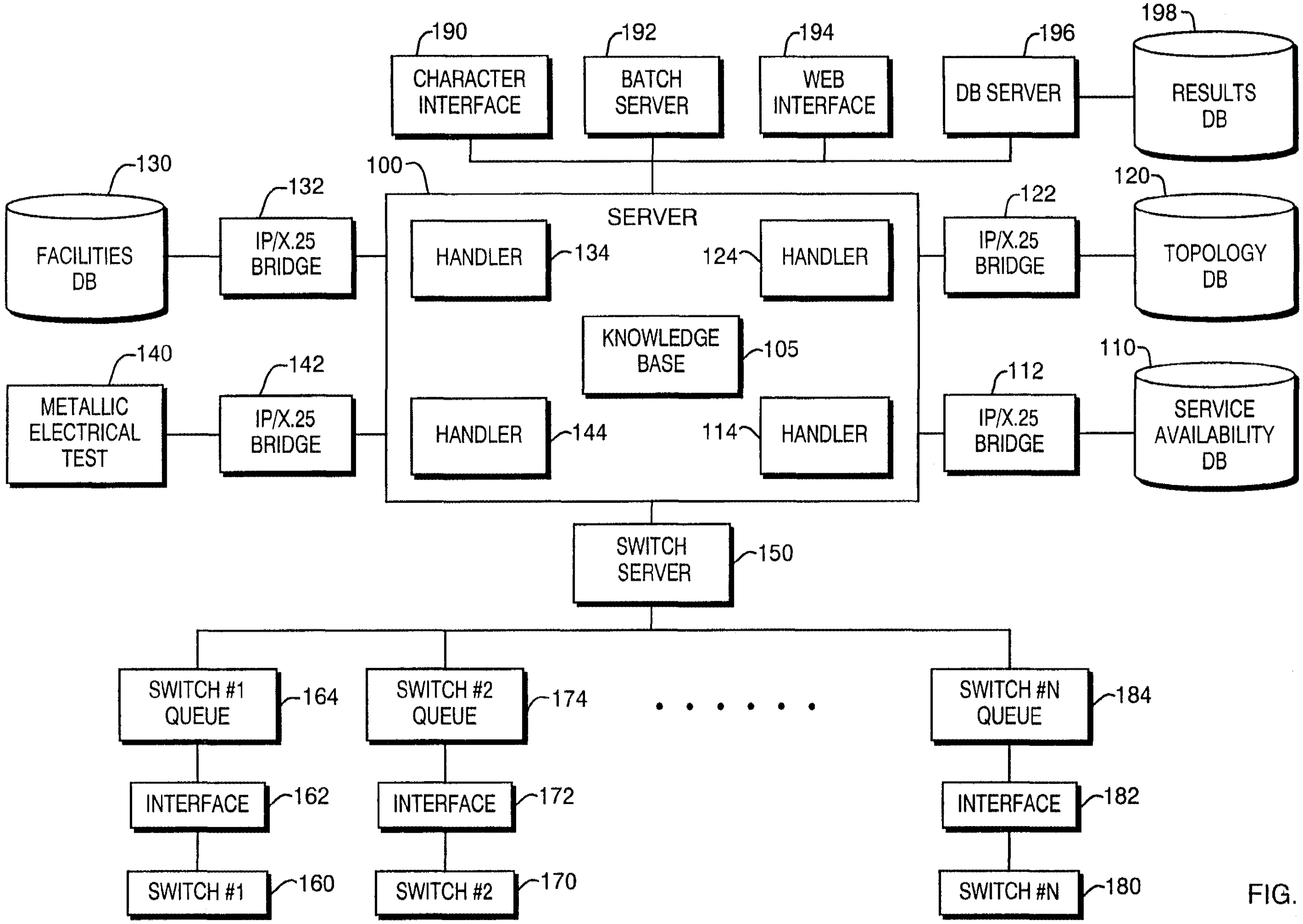
CHARACTER INTERFACE
190
BATCH SERVER
192
WEB INTERFACE
194
DB SERVER
196
RESULTS DB
198
FACILITIES DB
130
IP/X.25 BRIDGE
132
100
SERVER
HANDLER
134
124
HANDLER
IP/X.25 BRIDGE
122
TOPOLOGY DB
120
KNOWLEDGE BASE
105
METALLIC ELECTRICAL TEST
140
IP/X.25 BRIDGE
142
HANDLER
144
114
HANDLER
IP/X.25 BRIDGE
112
SERVICE AVAILABILITY DB
110
SWITCH SERVER
150
SWITCH #1 QUEUE
164
SWITCH #2 QUEUE
174
SWITCH #N QUEUE
184
INTERFACE
162
INTERFACE
172
INTERFACE
182
SWITCH #1
160
SWITCH #2
170
SWITCH #N
180

FIG. 4

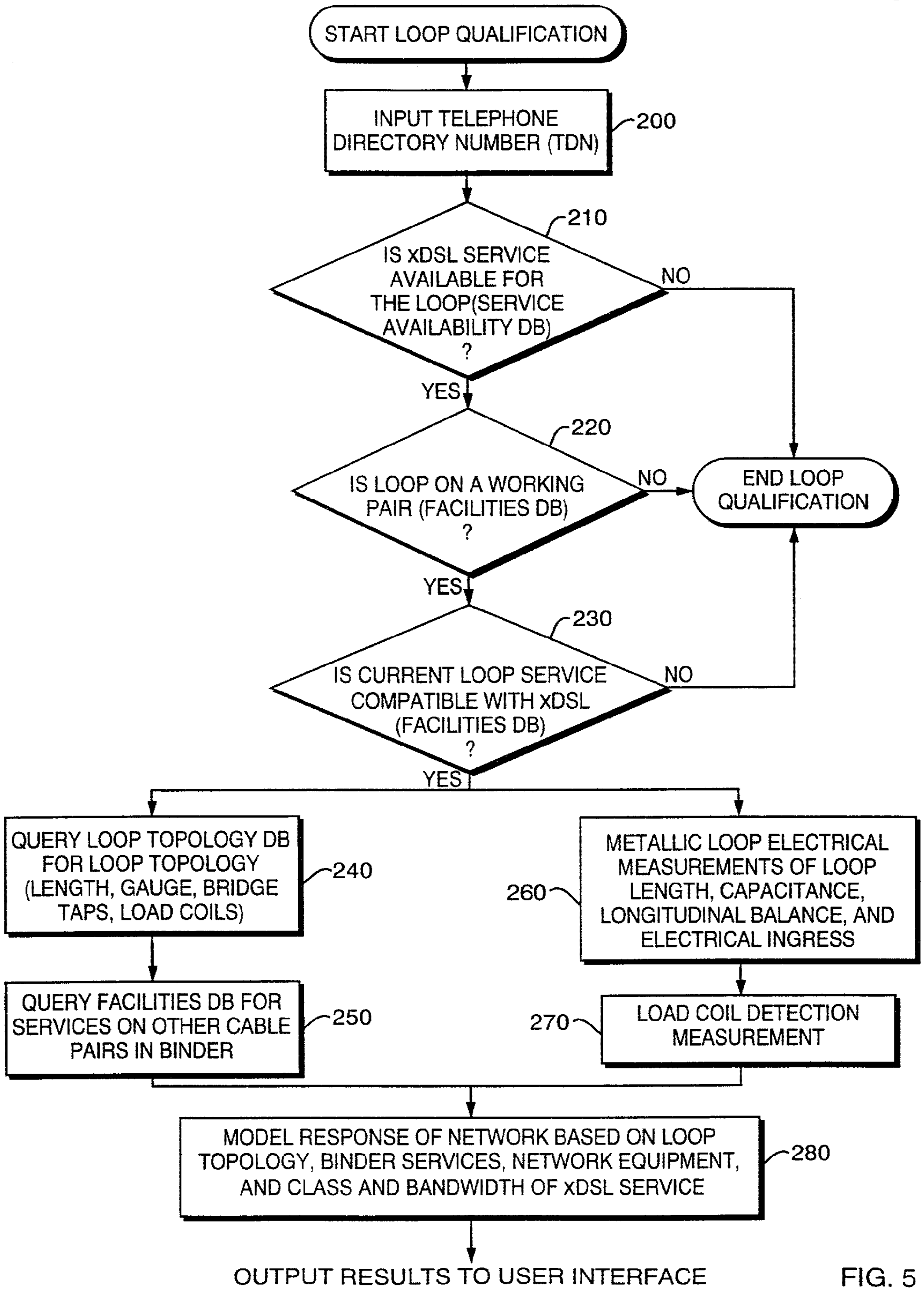
START LOOP QUALIFICATION
INPUT TELEPHONE DIRECTORY NUMBER (TDN)
200
210
IS xDSL SERVICE AVAILABLE FOR THE LOOP(SERVICE AVAILABILITY DB) ?
NO
YES
220
IS LOOP ON A WORKING PAIR (FACILITIES DB) ?
NO
END LOOP QUALIFICATION
YES
230
IS CURRENT LOOP SERVICE COMPATIBLE WITH xDSL (FACILITIES DB) ?
NO
YES
QUERY LOOP TOPOLOGY DB FOR LOOP TOPOLOGY (LENGTH, GAUGE, BRIDGE TAPS, LOAD COILS)
240
260
METALLIC LOOP ELECTRICAL MEASUREMENTS OF LOOP LENGTH, CAPACITANCE, LONGITUDINAL BALANCE, AND ELECTRICAL INGRESS
QUERY FACILITIES DB FOR SERVICES ON OTHER CABLE PAIRS IN BINDER
250
270
LOAD COIL DETECTION MEASUREMENT
MODEL RESPONSE OF NETWORK BASED ON LOOP TOPOLOGY, BINDER SERVICES, NETWORK EQUIPMENT, AND CLASS AND BANDWIDTH OF xDSL SERVICE
280
OUTPUT RESULTS TO USER INTERFACE

FIG. 5

INTERNATIONAL SEARCH REPORT	International application No. PCT/US99/11052

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :G06F 9/00; H04B 3/23, 3/32
US CL :702/57; 370/286, 294, 401; 375/296, 346, 349; 364/141
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 702/57; 370/286, 294, 401; 375/296, 346, 349; 364/141

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

APS

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5,181,198 A (LECHLEIDER) 19 January, 1993 (19.01.93), Abstract, figs.1-7, cols.1-3.	1-40
A	US 5,504,896 (SCHELL et al.) 02 April, 1996 (02.04.96), Abstract, figs.1-10, cols.1-2.	1-40

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

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Date of the actual completion of the international search	Date of mailing of the international search report
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Name and mailing address of the ISA/US Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231	Authorized officer HIEN VO
Facsimile No. (703) 305-3230	Telephone No. (703) 308-5253

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INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 : H04B 1/38, H04L 5/16, 27/10, 27/18	A1	(11) International Publication Number: WO 99/67890 (43) International Publication Date: 29 December 1999 (29.12.99)

(21) International Application Number: PCT/US99/13817

(22) International Filing Date: 18 June 1999 (18.06.99)

(30) Priority Data:
60/090,333 23 June 1998 (23.06.98) US

(71) Applicant *(for all designated States except US)*: PC TEL, INC. [US/US]; 70 Rio Robles, San Jose, CA 95134 (US).

(72) Inventors; and
(75) Inventors/Applicants *(for US only)*: GOLDSTEIN, Yuri [US/US]; 924 Bullet Hill Road, Southbury, CT 06488 (US). OKUNEV, Yuri [US/US]; 1023/B Heritage Village, Southbury, CT 06488 (US).

(74) Agent: GORDON, David, P.; 65 Woods End Road, Stamford, CT 06905 (US).

(81) Designated States: AU, CA, CN, JP, KR, US, European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

Published
With international search report.
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(54) Title: SPREAD SPECTRUM HANDSHAKE FOR DIGITAL SUBSCRIBER LINE TELECOMMUNICATIONS SYSTEMS

(57) Abstract

Handshake information for xDSL services is transmitted utilizing a spread spectrum modulated system where a plurality (n) of carrier tones ($n > 2$) are summed and utilized as a spread spectrum carrier (SSC), and data is modulated onto the carrier (at all utilized frequencies). Preferably, phase shift keying (PSK) modulation or a variation thereof is used as the encoding/modulation technique.

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RO	Romania
RU	Russian Federation
SD	Sudan
SE	Sweden
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SN	Senegal
SZ	Swaziland
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SPREAD SPECTRUM HANDSHAKE FOR DIGITAL SUBSCRIBER LINE TELECOMMUNICATIONS SYSTEMS

Priority is claimed from provisional application Serial No. 60/090,333 filed June 23, 1998 which is hereby incorporated by reference in its entirety herein.

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates broadly to telecommunications systems and methods. More particularly, the present invention relates to a handshake for an xDSL (Digital Subscriber Line type) modem.

2. State of the Art

Digital subscriber line (DSL) systems are a new and fast-growing data transmission service which provide significantly higher data rates than conventional V.34 and V.90 type modems. The abbreviation "xDSL" is an integrated designation for different DSL services including ADSL (asymmetric DSL), SDSL (symmetric DSL), RADSL (rate-adaptive DSL), HDSL (high speed DSL), and VDSL (very high speed DSL), UDSL (universal DSL), and their modifications such as ADSL-LITE (also known as G.lite). The xDSL services typically provide data rates of several Mbits/s downstream and several hundred Kbits/s upstream, although SDSL provides the same upstream and downstream rates. All types of DSL are based on discrete multitone (DMT) technology although they have different parameters. See, J. Makris, "DSL Services", Data Communications, April 1998, and ANSI T1.413 -1995 "Network and Customer Installation Interfaces - Asymmetrical Digital Subscriber Line (ADSL) Metallic Interface".

According to the ITU-T telecommunications standards for the xDSL services, at modem start-up a handshake procedure (called G.hs) is utilized. The requirements for G.hs are set forth in several documents such as "Proposal for G.hs Modulation

Technique and Message Protocol", ITU-T Telecommunication Standardization Sector, C1-068 Chicago, USA 6-9 April 1998, and "Handshake procedures for Digital Subscriber Line (DSL) transceivers", ITU-T Draft G.994.1 (February 3, 1999) which are both hereby incorporated by reference herein in their entireties. The main requirements of the handshake are: transmission of several tens of bytes during the handshake; signal compatibility with all types of DSL receivers; and interworking with the plain old telephone service (POTS), the integrated services digital network (ISDN), and time compression multiplexing ISDN (TCM-ISDN). Meeting these main requirements is not a trivial task because of considerable noise and cross-talk impairments, and lack of knowledge regarding the frequency characteristics of the channel, all of which is described in various papers such as: Matsushita Electric Industrial Co. Ltd, "Proposed Working Text for G.hs Based on V.8bis", ITU-Telecommunication Standardization Sector, NF-044, Nice, France, 11-14 May 1998; Matsushita Electric Industrial Co. Ltd., "Spectrum Considerations for G.hs", ITU-Telecommunications Standardization Sector, NF-045, Nice, France 11-14 May 1998; Matsushita Electric Industrial Co., Ltd., "Crosstalk Model Proposed Working Text for G.hs Test" ITU-Telecommunications Standardization Sector, NF-046, Nice, France 11-14 May 1998; NEC, "Desired Spectrum Range for G.hs under TCM-ISDN", ITU-Telecommunications Standardization Sector, NF-066, Nice, France 11-14 May 1998; and 3Com, "Proposed Spectrum and Tone Selection for G.hs", ITU-Telecommunications Standardization Sector, NF-068, Nice, France 11-14 May 1998.

More particularly, signal attenuation across lines carrying xDSL signals is a non-monotonic function of frequency, and may have several deep notches, while noise power spectral density (PSD) is also not a flat function of frequency. As a result, the signal to noise ratio (SNR) is a complex multiextremes function of frequency. Moreover, the SNR is subject to random and cyclic variations in time. For example, in the TCM-ISDN environment which includes the so-called "ping-pong mode" of up- and down-transmissions, far-end cross-talk (FEXT) and near-end cross-talk (NEXT) interleave at a frequency of 400 Hz. Since FEXT and NEXT processes have significantly different power spectral densities, significant NEXT noise is introduces every other 1.25 milliseconds.

As set forth above, several authors have made proposals regarding G.hs techniques. The core of these proposals has been two-tone transmission with different bit rates depending upon the noise environment. Frequency diversity is provided by bits duplication on nominal and backup carrier tones. Time diversity is provided by increasing the symbol interval (i.e., decreasing the symbol rate). These proposals have several disadvantages. First, both the nominal and backup tones may be located in notches or other frequency domain areas having a low SNR, thus rendering the handshake ineffective. Second, increasing the symbol interval may not be sufficient to account for bursty noise. For example, in the TCM-ISDN environment, the signal to noise ratio may be below an acceptable level every other 1.25 ms interval. Even if the initial symbol interval of .232 ms were to be increased by a factor of four to .928 ms as suggested by one of the authors in the art, the entire interval could be located within the 1.25 ms high noise window. In fact, even increasing the symbol interval by a factor of 8 would still only provide a final symbol interval of 1.885 ms which could be 67% covered by the low SNR area.

SUMMARY OF THE INVENTION

It is therefore an object of the invention to provide a handshake for an xDSL modem which meets proposed xDSL standards requirements.

It is another object of the invention to provide a handshake for an xDSL modem which has excellent frequency diversity and time diversity and provides excellent reliability.

It is a further object of the invention to provide an xDSL modem handshake which utilizes multitone signaling.

It is an additional object of the invention to provide an xDSL modem handshake which will interwork with existing telecommunications services.

Another object of the invention is to provide modems and methods for implementing the above-listed objects.

In accord with the objects of the invention, handshake information for xDSL services are transmitted utilizing a spread spectrum modulated system where a plurality (n) of carrier tones ($n > 2$) are summed and utilized as a spread spectrum carrier (SSC), and data is modulated onto the carrier (at all utilized frequencies). Preferably, phase shift keying (PSK) modulation (or a variation thereof such as BPSK - binary PSK, or DBPSK - differential binary PSK) is used as the modulating technique. When the spread spectrum carrier is modulated by handshake bits according to BPSK, the SSC is transmitted with sign "+" if the handshake bit is a +1 and with sign "-" if the handshake bit is a "-1". When using DPSK, the same modulation procedure is used for differentially encoded handshake bits.

According to one preferred aspect of the invention, the handshake symbol rate (SR) is set equal to .8A symbols/msec, where A is a positive integer. In order to improve reliability, symbols are preferably repeated at least four times. According to another preferred aspect of the invention, a preamble can be provided for timing recovery purposes. Further aspects of the invention include different receiver systems, including a quasicoherent receiver, an autocorrelation receiver, and a presently preferred incoherent receiver which utilizes coherent accumulation of FFT components for a DBPSK spread spectrum handshake signal.

Additional objects and advantages of the invention will become apparent to those skilled in the art upon reference to the detailed description taken in conjunction with the provided figures.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a block diagram of the preferred transmitter of the invention.

Fig. 2 is a diagram showing the signal structure of the preferred handshake signal of the invention.

Fig. 3a is a block diagram of an autocorrelation receiver of DBPSK signals according to the invention;

Fig. 3b is a block diagram of a quasicoherent receiver of DBPSK signals according to the invention; and

Fig. 3c is a block diagram of an incoherent receiver which utilizes coherent accumulation of FFT components for a DBPSK spread spectrum handshake signal according to the invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

According to the invention, handshake information for xDSL services is transmitted by modulating the handshake information on a spread spectrum carrier (SSC), where the SSC is a sum of tones conventionally used by xDSL for the data transmission mode. As seen in Fig. 1, the transmitter 10 includes a phase initialization (PI) unit 15, an inverse fast Fourier transformation (IFFT) unit 20, a spread spectrum carrier (SSC) memory 25, a modulator 30, a differential encoder 35 and a block frame unit 40. In essence, the phase initialization unit 15 generates complex numbers indicating a desirable amplitude and initial phase distribution for a plurality of multitone signals. Preferably, the amplitude distribution is chosen to be flat (uniform). According to the preferred embodiment, the initial phases of different tones are generated randomly or selected specifically in order to minimize the crest-factor of the generated tones. Regardless, where DMT-style implementation is utilized, the IFFT transforms a set of complex numbers into a set of time-domain samples which are stored in memory 25. If, for example, all or substantially all two hundred fifty-six DMT tones (such as might be utilized in ITU-T Standard G.992.2) are generated by the PI unit 15, a five hundred twelve sample set may be stored in the memory 25. Additional repetitive samples may also be stored in the memory, if desired as a prefix which can be used by the receiver to reduce distortion. If desired, the samples may be generated in other manners (e.g., without the PI and IFFT, or in another apparatus) and loaded and stored in the transmitter memory for use as described below.

While all two hundred fifty-six DMT tones may be included in the spread spectrum carrier, it should be appreciated by those skilled in the art that according to the invention, different numbers of tones (and different tones) can be used in different circumstances, provided a spread spectrum carrier is utilized. Thus, for purposes of this application, a carrier may be considered a spread spectrum carrier if three or more distinct tones are modulated together. Thus, the SSC for a down-stream connection may contain a full or partial set of down-stream tones, while the SSC for an up-stream connection could contain a full or partial set of up-stream tones. For example, a G.Lite ADSL up-stream SSC might utilize allowed tones from the set six through thirty-two (25.875 kHz... 138 KHz), while the downstream SSC might utilize allowed tones from the set thirty-three through one hundred twenty-eight (142.3125 kHz ... 552 kHz). The SSC may contain only even or odd tones to reduce the processing at the receiver.

Handshake information (as described below with reference to Fig. 2) which is to be modulated onto the spread spectrum carrier is provided to the differential encoder 35 and differentially encoded bits are written to the block frame unit 40. According to the preferred embodiment of the invention, the handshake information is provided to the differential encoder at a speed of .8 kbps, and differentially encoded 4-bit subblocks are written into registers of the block frame unit 40. Preferably, each 4-bit subblock is read four times such that each block frame is provided to the modulator 30 with a speed of 3.2 kbps.

When no differential encoder is utilized, the modulation technique is preferably is a binary phase shift keying (BPSK). When a differential encoder is utilized, the modulation technique is preferably a differential BPSK. Regardless, the modulator 30 uses the output of the block framer unit 40 to select whether the samples stored in the memory 25 are to be transmitted as is, or inverted (i.e., multiplied by -1 or 180 degrees out of phase). The samples stored in the memory 25 are sequentially read out of the memory so that all samples are modulated (i.e., transmitted as is or inverted) at the proposed symbol rate discussed below. When BPSK is utilized, the SSC samples are transmitted with sign "+" if the handshake bit is a "+1", and transmitted with sign "-" if the handshake bit is "-1" (or vice versa). When using DBPSK, the same modulation procedure is used for differentially encoded handshake bits.

It will be appreciated by those skilled in the art that while BPSK or DBPSK modulation is preferred, other modulation techniques such as QPSK (quadrature PSK), DQPSK (differential QPSK), frequency modulation, amplitude modulation, and quadrature amplitude modulation could be utilized.

Details of the handshake which modulates the SSC is seen in Fig. 2. According to the preferred embodiment of the invention, the handshake includes a preamble and a G.hs message. The preamble comprises N subblocks of a distinct four bit sequence "1,1,1,-1" followed by four subblocks of a four bit divider sequence "-1,-1,-1,-1", followed by eight subblocks of a pseudorandom sequence (as specified). Each subblock is preferably generated at a 1.25 millisecond rate (i.e., each subblock has a duration of 1.25 ms), with bits being generated at a .3125 millisecond rate. After the preamble, the G.hs message is provided and preferably includes N blocks which are generated at a 5 millisecond rate. Each block preferably includes four subblocks of four information bits (symbols) each (b1, b2, b3, b4), with the four information bits being repeated four times (i.e., each subblock within the block contains the same material). Each symbol carries one information bit. So each block of duration 5 milliseconds, carries four information bits with redundancy 3/4. As indicated in Fig. 2, each bit of the preamble and G.hs message is preferably modulated onto a spread spectrum carrier. As discussed in more detail below, the preamble is preferably provided to permit the receiver to detect G.hs transmission, to recover the spread spectrum carrier for coherent processing, and for symbol and block synchronization (timing recovery). While the preamble is preferably modulated, an unmodulated preamble (all +1s) can be utilized.

According to the preferred embodiment of the invention, a symbol rate (SR) is set equal to .8 symbols/millisecond, where A = 1,2,3... With the symbol rate set in this manner, an integer number of symbols will be placed within the 1.25 millisecond burst duration in the TCM-ISDN cross talk environment. Thus, when A=4 (bit rate = 3200 bps), half a byte (four bits) will be transmitted within the 1.25 ms burst. When A=8 (bit rate = 6400 bps), one byte will be transmitted within the 1.25 ms burst. By transmitting each symbol of the G.hs message at least four times, at least two symbol time-separated blocks will occur within the 1.25 ms high SNR FEXT areas in a TCM-ISDN cross-talk environment.

Taking into account the 400 Hz periodicity of the NEXT and FEXT cross-talk in TCM-ISDN systems, a noiseless time window may be found by calculating the correlation between N-symbol blocks delayed by 2.5 ms relative to each other. If the delayed blocks coincide with each other (i.e., they have not been corrupted by noise), the time window has a "high enough" SNR (i.e., it is "noiseless" for the purpose of the handshake) and can be used for receiving the handshake message. The structure of the preamble is particularly arranged to permit this determination.

Because the noiseless time window has a random time position relative to the transmission of the preamble and handshake message, received N-symbol blocks may be cyclically shifted. In other words, the block frame may not correspond to the noiseless time window frame. It is therefore preferred that this shift be estimated and eliminated. According to the preferred embodiment of the invention, the cyclic shift may be estimated and eliminated by transmitting an N-symbol reference block. Thus, the preamble is provided with a series of reference blocks having the form "1,1,1,-1". It should be appreciated that any shift of the reference block will be distinct (-1,1,1,1; 1,-1,1,1; 1,1,-1,1) and detectable, and may therefore be detected and eliminated at the receiver. This pattern therefore allows for symbol synchronization and subblock synchronization.

Turning now to Figs. 3a-3c, three different receivers are shown for receiving and demodulating the handshake signal of the invention. An autocorrelation receiver 100a for DBPSK spread spectrum handshake signals is seen in Fig. 3a. The autocorrelation receiver 100a includes an autocorrelation demodulator 102a, a timing signal extractor 103a, and preferably further includes a noiseless time window (TW) determination unit 104a and a transmitted bit selection (BS) unit 106a. The autocorrelation demodulator 102a includes a delay line (DL) 110a, a multiplier 112a, a low pass filter (LPF) 114a, and a binary slicer (Sgn) 116a. Incoming SSC modulated signals are provided to the delay line 110a and the multiplier 112a. The delay Δt of the delay line is preferably set equal to 1/.8A ms (i.e., the handshake symbol duration). Thus, the multiplier 112a multiplies the incoming signal with the delayed signal. The output is forwarded to the low pass filter 114a which is preferably provided with a frequency bandwidth Δf approximately equal to A/1.25 kHz. For example, when using block length A=4, Δt = 0.3125 ms, and Δf = 3.2 kHz. The output of the low pass filter 114a reflects the modulation function in the transmitter, and the sign function of the low pass filter output, as generated by the binary slicer 116a which compares the output to a zero threshold, corresponds to the transmitted bits.

As will be appreciated by those skilled in the art, the autocorrelation receiver 100a calculates (at the multiplier 112a) a scaler product ($S_n(t)*S_{n-1}(t)$) between a given spread spectrum signal $S_n(t)$ and a previous spread spectrum signal $S_{n-1}(t)$. The binary symbol I_n received with the n-th symbol interval is therefore determined according to $I_n = \mathrm{sgn}(S_n(t)*S_{n-1}(t)$.

As seen in Fig. 3a, the binary slicer 116a requires timing information which is preferably extracted from the low pass filter output by bandpass filtering of a frequency component responding to the baud (symbol) frequency. Alternatively (and also as shown in Fig. 3a), the timing information can be extracted from the incoming signal by a variety of well-known methods; e.g., as taught in Jan W. M. Bergmans, Digital Baseband Transmission and Recording, Chapters 9 and 10, "Basics of Timing Recovery", and "A Catalog of Timing Recovery Schemes", Kluwer Academic Publishers, Boston (1996) pp. 451-587.

While the autocorrelation demodulator 102a in conjunction with the timing extractor 103a suffices as a G.hs receiver in situations which do not require carrier recovery or other special synchronization, additional circuitry can be utilized if desired. Thus, is the channel noise has a steady power spectral density, robustness can be increased by accumulating signals at the output of the low pass filter, taking into account that every symbol may be repeated several times. In addition, if the PSD is known, the spread spectrum signal may be passed through a corresponding filter (not shown) at the input of the receiver in order to emphasize components of the spread spectrum signal having a higher SNR.

In addition, and according to the preferred embodiment of the invention, where a preamble is utilized, a noiseless time window determination unit 104a can be provided to compare the signal subblocks containing N symbols and delayed relative to each other by 2.5 ms. If the delayed N bit combination coincides within a certain time window, it indicates that this window has a sufficiently high SNR and can be used for receiving handshake bits. Regardless, the window determination unit 104 finds the time window of interest and generates an output signal indicating the time position of the desired window which is provided to the bit selection unit 106a. The demodulated bits provided at the output of the slicer during the noiseless window are also provided to the bit selection unit 106a, which determines from the bits and the window information the cyclic shift in effect. Thus, during receipt of the G.hs message, the bit selection unit 106a selects the correct portion of the received bits and eliminates the cyclic shift in the received information blocks. The bit selection unit 106a produces for output N bits every 5 milliseconds.

Turning to Fig. 3b, a quasicoherent receiver 100b for DBPSK spread spectrum handshake signals is shown. The quasicoherent receiver 100b includes an autocorrelation demodulator 102b, a timing signal extractor 103b, and preferably further includes a noiseless time window determination unit 104b and a transmitted bit selection unit 106b. The quasicoherent demodulator 102b includes a spread spectrum recovery (SSCR) unit 111b, a multiplier 112b, a low pass filter 114b, a binary slicer 116b, a delay line 118b, and a sign multiplier 120b. Incoming SSC modulated signals are provided to the spread spectrum carrier recovery unit 111b and the multiplier 112b. The spread spectrum carrier recovery unit 111b accumulates SSC samples during the preamble and extracts a spread spectrum reference signal R(t) therefrom. The multiplier 112b multiplies the incoming signal with the output of the SSC recovery unit. The output is forwarded to the low pass filter 114b which is preferably provided with a frequency bandwidth Δf approximately equal to N/1.25 kHz. The output of the low pass filter 114b is fed to slicer 116b which compares the output to a threshold (typically zero). The output of slicer 116b is a binary signal which is fed to the delay line 118b and to the sign multiplier 120b. The sign of the output of the sign multiplier 120b corresponds to the transmitted bits.

As will be appreciated by those skilled in the art, in the quasicoherent receiver 100b, the average unmodulated SSC, preferably extracted from the preamble by the SSC recovery unit 111b, is used as a spread spectrum reference signal R(t) for the coherent demodulation. Thus, the recovered binary symbol $I_n = J_n * J_{n-1}$, where $J_n = \mathrm{sgn}(S_n(t) * R(t))$, and $J_{n-1} = \mathrm{sgn}(S_{n-1}(t) * R(t))$. The quasicoherent receiver 100b provides excellent results, but is substantially more complicated to implement than the autocorrelation receiver 100a because of the SSC recovery unit 111b.

The functioning of the timing signal extractor 103b, and the time window determination unit 104b and bit selection unit 106b of the quasicoherent receiver 100b are substantially as described above with respect to corresponding elements of Fig. 3a.

Turning to Fig. 3c, an incoherent receiver 100c for DBPSK spread spectrum handshake signals is shown. As seen in Fig. 3c, the incoherent receiver includes a fast Fourier transform block 130, a quadrature component accumulation unit 135, a multichannel incoherent demodulator 140, a DMT accumulation unit 145, and a binary slicer 150. The FFT block 130 receives the time domain handshake signal and converts the signal into a frequency domain signal. The output of the FFT block are signals F_{cnkm} and F_{snkm} which are respectively, the real and complex parts for the k-th DMT tone at the m-th DMT symbol interval of the n-th handshake symbol. The quadrature component accumulation (QCA) unit 145 separately sums the real parts together and the imaginary parts together according to $F_{cnk} = \sum_m F_{cnkm}$ and $F_{snk} = \sum_m F_{snkm}$. The outputs of the quadrature component accumulation unit 145 are then demodulated by the incoherent demodulator 140 according to $F_{nk} = F_{cnk} * F_{c(n-1)k} + F_{snk} * F_{s(n-1)k}$. The outputs of the incoherent demodulator 140 are then summed over all tones k by the DMT accumulator (DMTA) 145 according to $F_n = \sum_k F_{nk}$. Finally, the output of the DMT accumulator 145 is provided to the binary slicer 150 in order to compare the output F_n to a zero threshold. The decoded binary symbol $I_n = \mathrm{sgn}(F_n)$.

It should be appreciated by those skilled in the art that the incoherent receiver 100c is relatively simple to implement because it is based on the use of a FFT which is already available in DMT-based systems. In addition, no frequency equalization (carrier phase recovery) is required, and the performance of the incoherent receiver 100c is nearly as good as the quasicoherent receiver 100b of Fig. 3b.

There have been described and illustrated herein methods and apparatus for implementing a spread spectrum handshake for a digital subscriber line telecommunications system. While particular embodiments of the invention have been described, it is not intended that the invention be limited thereto, as it is intended that the invention be as broad in scope as the art will allow and that the specification be read likewise. Thus, while a particular transmitter and particular receivers have been disclosed, it will be appreciated that other transmitters and receivers could be utilized, provided that the transmitter modulate a handshake signal onto a spread spectrum carrier. Thus the implementation of the transmitters and receivers will partially depend upon the encoding technique utilized (e.g., DPSK, QPSK, etc.), the results desired, and limitations or requirements of standards which might be applicable. Implementation of functions may also be accomplished in several manners. Thus, while slicers have been described for purposes of generating decoded binary signals, other apparatus well-known in the art could be utilized. Also, while a handshake sequence including a preamble and a handshake message have been described, it will be appreciated that different preambles and different handshake messages could be provided, and/or that a handshake sequence could be provided with no preamble. It will therefore be appreciated by those skilled in the art that yet other modifications could be made to the provided invention without deviating from its spirit and scope as so claimed.

We claim:

1. A digital subscriber line (DSL) type modem, comprising:
a transmitter having
a handshake generator which generates handshake signals,
a spread spectrum carrier generator which generates a spread spectrum carrier including at least three tones associated with DSL type modems, and
a modulator coupled to said handshake generator and to said spread spectrum carrier generator, said modulator modulating indications of said handshake signals onto indications of said spread spectrum carrier simultaneously.

2. A modem according to claim 1, wherein:
said modulator modulates said indications of said spread spectrum carrier according to one of a phase shift keying (PSK) technique, frequency modulation, amplitude modulation, and quadrature amplitude modulation.

3. A modem according to claim 2, wherein:
said PSK technique comprises one of binary PSK, differential binary PSK, quadrature PSK, and differential quadrature PSK.

4. A modem according to claim 1, wherein:
said modulator modulates said indications of said spread spectrum carrier according to differential binary phase shift keying.

5. A modem according to claim 4, wherein:
said spread spectrum carrier generator comprises memory which stores said indications of all said tones.

6. A modem according to claim 5, wherein:
said indications comprise inverse fast Fourier transform (IFFT) samples of said at least three tones.

7. A modem according to claim 6, wherein:
said indications comprise IFFT samples of substantially all two hundred fifty-six DMT tones associates with DSL type modems.

8. A modem according to claim 1, wherein:
said spread spectrum carrier generator comprises memory which stores said indications of all said tones.

9. A modem according to claim 8, wherein:
said indications of all said tones comprise inverse fast Fourier transform (IFFT) samples of said at least three tones.

10. A modem according to claim 9, wherein:
said indications of all said tones comprise IFFT samples of substantially all two hundred fifty-six DMT tones associates with DSL type modems.

11. A modem according to claim 1, wherein:
said handshake generator comprises a differential encoder coupled to a block framer.

12. A modem according to claim 1, wherein:
said handshake signals comprise a handshake message.

13. A modem according to claim 12, wherein:
said handshake message includes a plurality of blocks, each block having a plurality of repeating subblocks.

14. A modem according to claim 13, wherein:
said blocks have a 5 millisecond rate.

15. A modem according to claim 14, wherein:
said subblocks have a 1.25 millisecond rate, and each subblock contains four bits.

16. A modem according to claim 12, wherein:
said handshake signals further comprise a preamble.

17. A modem according to claim 16, wherein:
said preamble comprises a plurality of repeating subblocks.

18. A modem according to claim 17, wherein:
each said subblock has a 1.25 millisecond rate and includes four predetermined bits, said four predetermined bits selected to permit a shift in phase of said four predetermined bits to be detected.

19. A modem according to claim 17, wherein:
said preamble further includes at least one subblock having a divider sequence, and a plurality of subblocks representing a pseudorandom sequence.

20. A modem according to claim 1, further comprising:
a receiver having a demodulator.

21. A modem according to claim 20, wherein:
said receiver is chosen from a group consisting of an autocorrelation receiver, a quasicoherent receiver, and an incoherent receiver.

22. A modem according to claim 21, wherein:
said receiver is an autocorrelation receiver including a delay line which receives and delays a received handshake signal, a multiplier which multiplies said received handshake signal with an output of said delay line, a low pass filter which filters an output of the multiplier, and means for obtaining binary symbol indication from an output of said low pass filter.

23. A modem according to claim 21, wherein:
said receiver is a quasicoherent receiver including a spread spectrum carrier recovery unit generates a reference spread spectrum signal from the received signal, a multiplier which multiplies a received signal with said reference signal, a low pass filter which filters an output of the multiplier, and means for obtaining a binary symbol indication from an output of said low pass filter.

24. A modem according to claim 23, wherein:
said means for obtaining a binary symbol indication comprises a slicer coupled to an output of said low pass filter, a delay line which receives and delays outputs of said slicer, and a second multiplier which receives an output of said slicer and an output of said delay line and generates a binary symbol indication therefrom.

25. A modem according to claim 21, wherein:
said receiver is an incoherent receiver including a fast Fourier transformer (FFT) which receives an incoming time domain handshake signal and generates real and imaginary frequency domain signals therefrom, a quadrature component accumulation (QCA) unit coupled to said FFT which separately sums said real frequency domain signals together and said imaginary frequency domain signals together, an incoherent demodulator coupled to said QCA unit which combines said summed real and imaginary frequency domain signals, a discrete multitone accumulator (DMTA) coupled to said QCA unit which sums outputs of said QCA unit over said at least three tones, and means for generating a decoded binary symbol from an output of said DMTA.

26. A modem according to claim 20, wherein:
said handshake signals comprise a handshake message and a preamble, said preamble comprises a plurality of repeating subblocks, wherein said receiver includes means for utilizing said repeating subblocks to find a high-signal-to-noise time window.

27. A modem according to calim 26, wherein

said means for utilizing said repeating subblocks includes means for correlation of said repeating subblocks delayed relative to each other by a predetermined time interval.

28. A method of transmitting digital subscriber line (DSL) type modem handshake information, comprising:

generating handshake signals; and

modulating indications of said handshake signals onto a spread spectrum carrier, said spread spectrum carrier including at least three tones associated with DSL type modems, wherein said modulating comprises modulating said indications of said handshake signals onto indications of said at least three tones simultaneously.

29. A method according to claim 28, wherein:

said handshake signal indications are modulated onto said spread spectrum carrier according to one of a phase shift keying (PSK) technique, frequency modulation, amplitude modulation, and quadrature amplitude modulation.

30. A method according to claim 29, wherein:

said PSK technique comprises one of binary PSK, differential binary PSK, quadrature PSK, and differential quadrature PSK.

31. A method according to claim 28, wherein:

said handshake signal indications are modulated onto said spread spectrum carrier according to differential binary phase shift keying.

32. A method according to claim 28, further comprising:

generating said indications by taking an inverse fast Fourier transform (IFFT) of said at least three tones; and

storing said indications in memory, wherein said modulating comprises reading said indications from memory in order to modulate said indications of said handshake signals onto said indications stored in memory.

33. A method according to claim 28, wherein:
said handshake signals comprise a handshake message, said handshake message including a plurality of blocks, each block having a plurality of repeating subblocks.

34. A method according to claim 33, wherein:
said blocks have a 5 millisecond rate, said subblocks have a 1.25 millisecond rate, and each subblock contains four bits.

35. A method according to claim 33, wherein:
said handshake signals further comprise a preamble.

36. A method according to claim 33, wherein:
said preamble comprises a plurality of repeating subblocks, each said subblock has a 1.25 millisecond rate and includes four predetermined bits, said four predetermined bits selected to permit a shift in phase of said four predetermined bits to be detected.

37. A method according to claim 36, wherein:
said preamble further includes at least one subblock having a divider sequence, and a plurality of subblocks representing a pseudorandom sequence.

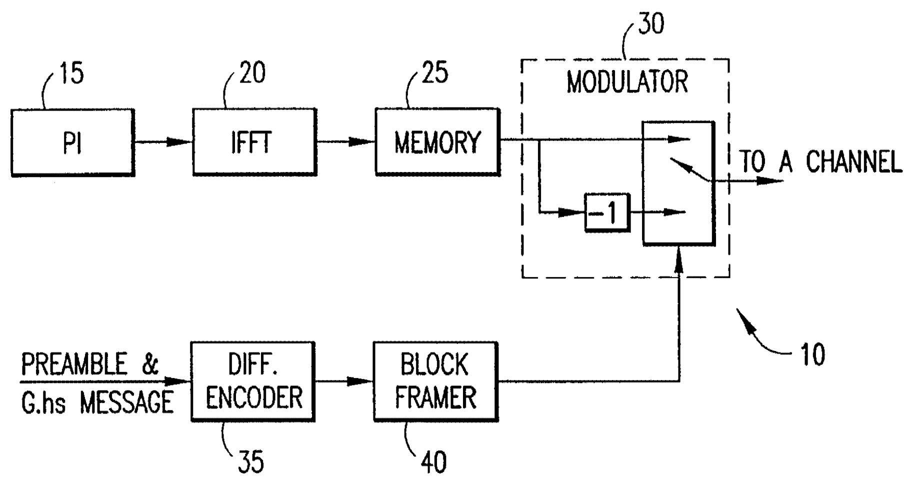
15
PI
20
IFFT
25
MEMORY
30
MODULATOR
−1
TO A CHANNEL
10
PREAMBLE &
G.hs MESSAGE
DIFF.
ENCODER
35
BLOCK
FRAMER
40

FIG.1

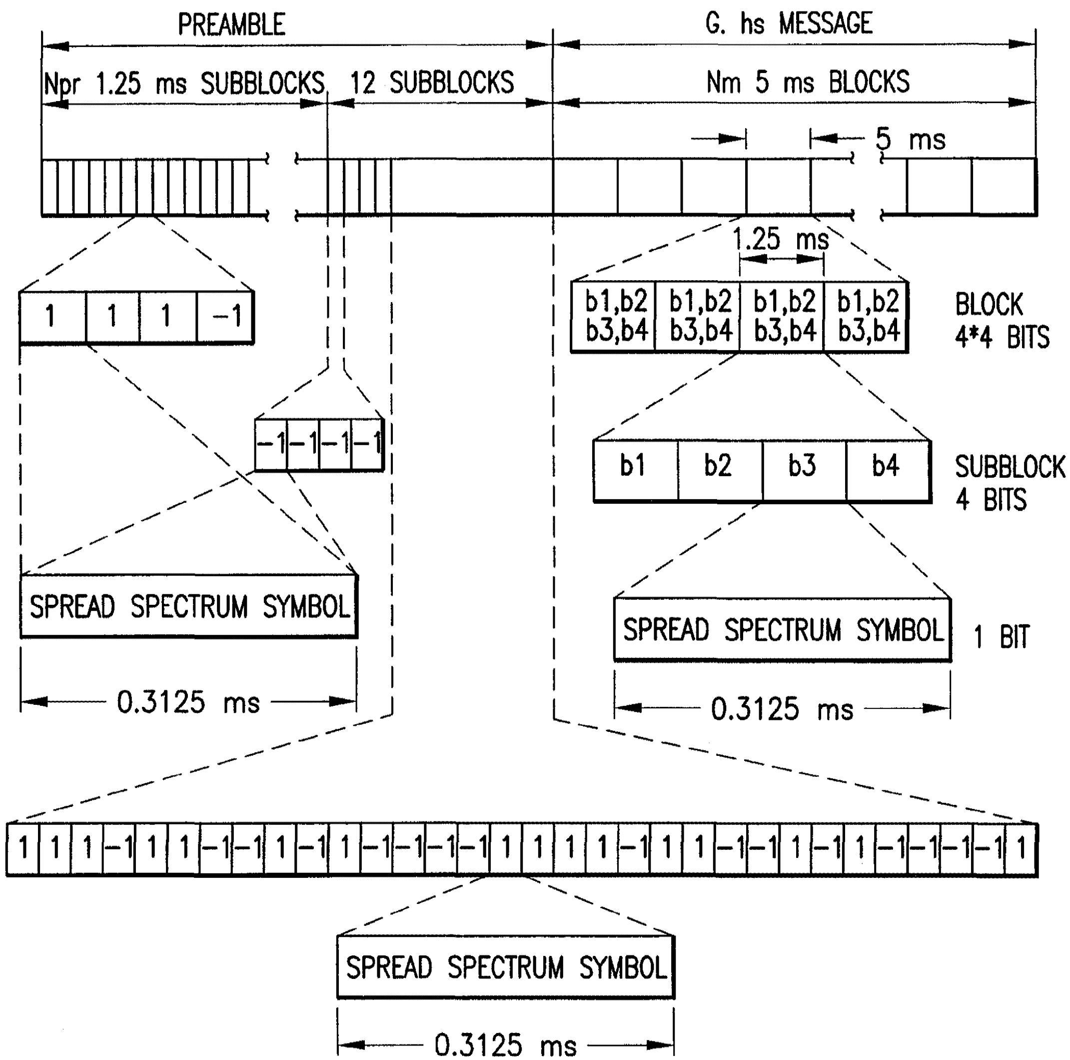
PREAMBLE
G. hs MESSAGE
Npr 1.25 ms SUBBLOCKS
12 SUBBLOCKS
Nm 5 ms BLOCKS
5 ms
1.25 ms
1
1
1
-1
-1
-1
-1
-1
b1,b2 b3,b4
b1,b2 b3,b4
b1,b2 b3,b4
b1,b2 b3,b4
BLOCK 4*4 BITS
b1
b2
b3
b4
SUBBLOCK 4 BITS
SPREAD SPECTRUM SYMBOL
0.3125 ms
SPREAD SPECTRUM SYMBOL
1 BIT
0.3125 ms
1 1 1 -1 1 1 -1 -1 1 -1 1 -1 -1 -1 -1 1 1 1 1 -1 1 1 -1 -1 1 -1 1 -1 -1 -1 -1 1
SPREAD SPECTRUM SYMBOL
0.3125 ms

FIG.2

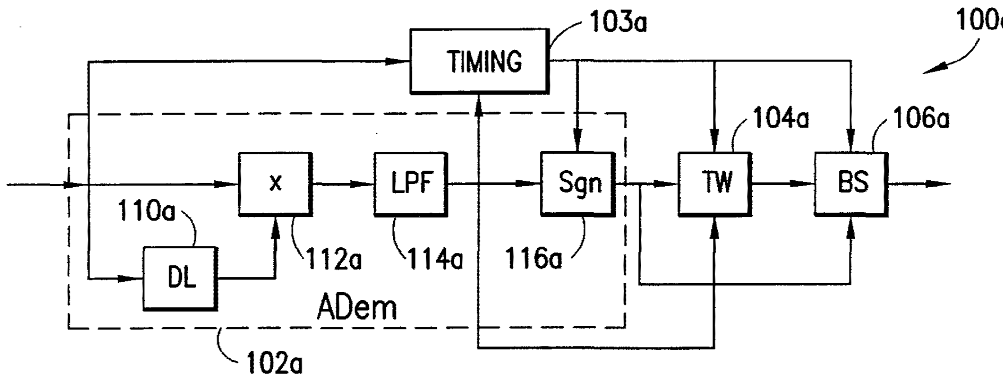

FIG.3a

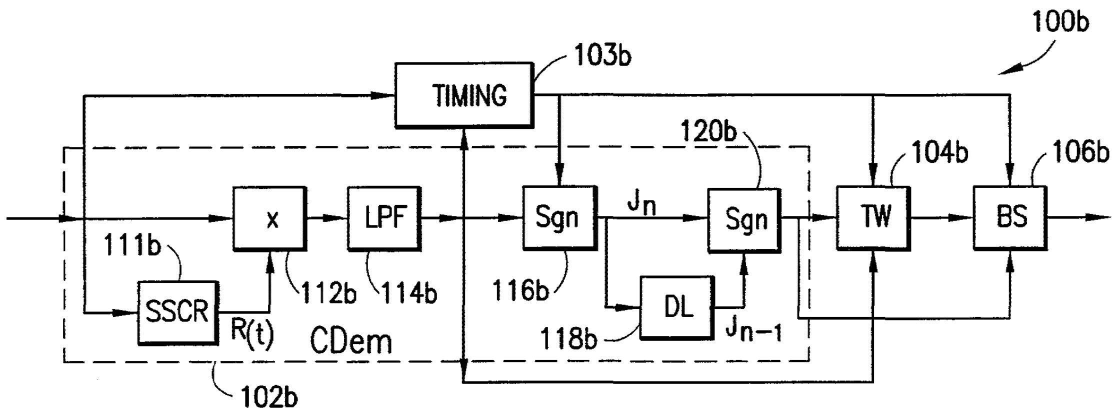

FIG.3b

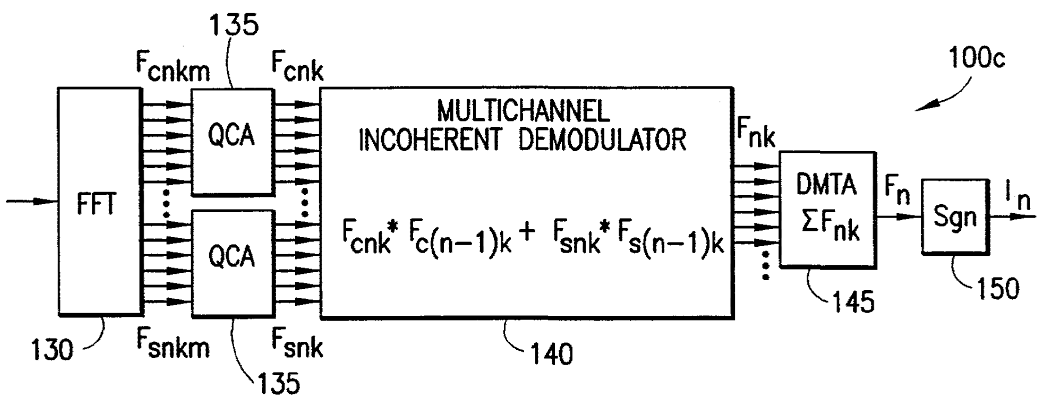

FIG.3c

SUBSTITUTE SHEET (RULE 26)

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US99/13817

A. CLASSIFICATION OF SUBJECT MATTER
IPC(6) :H04B 1/38; H04L 5/16, 27/10, 27/18
US CL :375/222, 223, 200, 208, 209, 210
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
U.S. : 375/222, 223, 200, 208, 209, 210

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
APS

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 5,751,701 A (LANGBERG et al.) 12 May 1998, Figs. 3,4,7, and 10, cols. 3-7.	1-10, 12-21, 26-37
Y,P	US 5,909,463 A (JOHNSON et al.) 01 June 1999, Figs. 1-4, col. 1, lines 20-60 and col. 16, lines 15-64.	1-10, 12-21, 26-37
Y	US 5,644,573 A (BINGHAM et al.) 01 July 1997, col. 8, lines 50-60.	3-7, 9, 10, 31 and 32
A,P	US 5,883,907 A (HOEKSTRA) 16 March 1999, Fig. 2 and col. 3, lines 27-35.	1-37

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search 23 AUGUST 1999	Date of mailing of the international search report 21 OCT 1999
Name and mailing address of the ISA/US Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231 Facsimile No. (703) 305-3230	Authorized officer MICHAEL W. MADDOX Telephone No. (703) 308-9557

Form PCT/ISA/210 (second sheet)(July 1992)★

81513-8

PATENT COOPERATION TREATY

From the
INTERNATIONAL PRELIMINARY EXAMINING AUTHORITY

PCT

RECEIVED JAN 24 2002 NIXON PEABODY

To:
Vick, Jason H.
Nixon Peabody LLP
8180 Greensboro Drive, Suite 800
McLean, Virginia 22102
ETATS-UNIS D'AMERIQUE

WRITTEN OPINION

(PCT Rule 66)

Date of mailing (*day/month/year*) 18/01/2002		
Applicant's or agent's file reference 081513.4	**REPLY DUE** within 1 / 00 months/days from the above date of mailing	
International application No. PCT/US 01/ 00418	International filing date (*day/month/year*) 08/01/2001	Priority date (*day/month/year*) 07/01/2000
International Patent Classification (IPC) or both national classification and IPC H04L1/24		
Applicant AWARE, INC.		

1. This written opinion is the first drawn up by this International Preliminary Examining Authority.
2. This opinion contains indications relating to the following items:
   - I [X] Basis of the opinion
   - II [ ] Priority
   - III [X] Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   - IV [ ] Lack of unity of invention
   - V [X] Reasoned statement under Rule 66.2(a)(ii) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement
   - VI [ ] Certain documents cited
   - VII [ ] Certain defects in the international application
   - VIII [ ] Certain observations on the international application
3. The applicant is hereby **invited to reply** to this opinion.

   When? See the time limit indicated above. The applicant may, before the expiration of that time limit, request this Authority to grant an extension, see Rule 66.2(d).

   How? By submitting a written reply, accompanied, where appropriate, by amendments, according to Rule 66.3. For the form and the language of the amendments, see Rules 66.8 and 66.9.

   Also For an additional opportunity to submit amendments, see Rule 66.4. For the examiner's obligation to consider amendments and/or arguments, see Rule 66.4*bis*. For an informal communication with the examiner, see Rule 66.6.

   If no reply is filed, the international preliminary examination report will be established on the basis of this opinion.
4. The final date by which the international preliminary examination report must be established according to Rule 69.2 is: 07/05/2002 .

Name and mailing address of the IPEA/ European Patent Office D-80298 Munich Tel. (+49-89) 2399-0, Tx: 523656 epmu d Fax: (+49-89) 2399-4465	Authorized officer Examiner Formalities officer (incl. extension of time limits) Tel. (+49-89) 2399 2828

EUROPÄISCHES PATENTAMT · EUROPEAN PATENT OFFICE · OFFICE EUROPÉEN DES BREVETS

Form PCT/IPEA/408 (cover sheet) (July 1998)

DOCKETED 1/24/02 By JM
Nixon Peabody, LLP

I. Basis of the opinion

1. The basis of this written opinion is the application as originally filed.

III. Non-establishment of opinion with regard to novelty, inventive step and industrial applicability

2. The question of whether the claimed invention appears to be novel, to involve an inventive step, or to be industrially applicable has not been and will not be the subject of the international preliminary examination (Article 34 (4) (a) (i) (ii) PCT; see also international search report) in respect of:

2.1 Applications having an unnecessary plurality of independent claims (generally not more than 1 independent claim in the same category is necessary; Article 6 PCT);

2.2 unsearched subject-matter (Article 17 (2) (a), Rule 66.1 (e) PCT), e.g.

2.2.1 claimed subject-matter under Rule 39.1 PCT,

2.2.2 applications where the description, the claims, or the drawings fail to comply with the prescribed requirements to such an extent that no meaningful search could have been carried out;

2.3 claimed subject-matter under Rule 67.1 PCT.

V. Reasoned statement under Rule 66.2(a)(ii) with regard to novelty, inventive step or industrial applicability

3. To the extent that the international preliminary examination has been carried out (see item III above), the following is pointed out:

In light of the documents cited in the international search report, it is considered that the invention as claimed in at least one of the independant claims does not appear to meet the criteria mentioned in Article 33 (1) PCT, i.e. does not appear to be novel and/or to involve an inventive step.

4. If amendments are filed, the Applicant must comply with the requirements of Rule 66.8 PCT and indicate the basis in the originally filed application of the amendments made (Article 34 (2) (b) PCT) otherwise these amendments will not be taken into consideration for the establishment of international preliminary examination.
The attention of the applicant is drawn to the fact that if the application contains an unjustified plurality of independent claims no examination of any of the claims will be carried out.

PATENT COOPERATION TREATY

PCT

REC'D 15 MAR 2002

PCT

INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

7

Applicant's or agent's file reference 081513.4	FOR FURTHER ACTION	See Notification of Transmittal of International Preliminary Examination Report (Form PCT/IPEA/416)
International application No. PCT/US 01/ 00418	International filing date (*day/month/year*) 08/01/2001	Priority date (*day/month/year*) 07/01/2000
International Patent Classification (IPC) or national classification and IPC H04L1/24		
Applicant AWARE, INC.		

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.

2. This REPORT consists of a total of 2 sheets, including this cover sheet.

   ☐ This report is also accompanied by ANNEXES, i.e., sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

   These annexes consists of a total of ______ sheets.

3. This report contains indications relating to the following items:

   - I ☒ Basis of the report
   - II ☐ Priority
   - III ☒ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   - IV ☐ Lack of unity of invention
   - V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement
   - VI ☐ Certain documents cited
   - VII ☐ Certain defects in the international application
   - VIII ☐ Certain observations on the international application

Date of submission of the demand	Date of completion of this report
02/08/2001	09/03/2002
Name and mailing address of the IPEA/ European Patent Office D-80298 Munich Tel. (+49-89) 2399-0, Tx: 523656 epmu d Fax: (+49-89) 2399-4465	Authorized officer VAN DEN BERG J G J Tel. (+49-89) 2399 2828

EUROPÄISCHES PATENTAMT · EUROPEAN PATENT OFFICE · OFFICE EUROPÉEN DES BREVETS

Form PCT/IPEA/409 (cover sheet) (July 1998)

I. Basis of the report

1. The basis of this international preliminary examination report is the application as originally filed.

III. Non-establishment of opinion with regard to novelty, inventive step and industrial applicability

2. The question of whether the claimed invention appears to be novel, to involve an inventive step, or to be industrially applicable has not been and will not be the subject of the international preliminary examination (Article 34 (4) (a) (i) (ii) PCT; see also international search report) in respect of:

2.1 Applications having an unnecessary plurality of independent claims (generally not more than 1 independent claim in the same category is necessary; Article 6 PCT);

2.2 unsearched subject-matter (Article 17 (2) (a), Rule 66.1 (e) PCT), e.g.

2.2.1 claimed subject-matter under Rule 39.1 PCT,

2.2.2 applications where the description, the claims, or the drawings fail to comply with the prescribed requirements to such an extent that no meaningful search could have been carried out;

2.3 claimed subject-matter under Rule 67.1 PCT.

V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement

3. To the extent that the international preliminary examination has been carried out (see item III above), the following is pointed out:

In light of the documents cited in the international search report, the invention as claimed in at least one of the independant claims does not appear to meet the criteria mentioned in Article 33 (1) PCT, i.e. does not appear to be novel and/or to involve an inventive step.

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 01/41653

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04B3/46 H04L27/26

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04L H04B

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, PAJ, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 6 075 821 A (CHEN CHUNTA ET AL) 13 June 2000 (2000-06-13) abstract column 6, line 24 - line 29 column 11, line 4 - line 14 claim 1	1,2
A	---	3-57
A	WO 99 63427 A (GTE LABORATORIES INC) 9 December 1999 (1999-12-09) the whole document ---	1-57
A	EP 0 889 615 A (INTEGRATED TELECOM EXPRESS) 7 January 1999 (1999-01-07) abstract page 4, line 12 - line 13 page 4, line 53 -page 5, line 3 claims 40-53 --- -/--	1-57

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
26 September 2002	09/10/2002

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Reilly, D

Form PCT/ISA/210 (second sheet) (July 1992)

INTERNATIONAL SEARCH REPORT

Internatic ication No
PCT/US 01/41653

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
A	US 5 128 619 A (BJORK ROGER A ET AL) 7 July 1992 (1992-07-07) the whole document -----	1-57

Form PCT/ISA/210 (continuation of second sheet) (July 1992)

INTERNATIONAL SEARCH REPORT

Internation ication No
PCT/US 01/41653

Patent document cited in search report		Publication date	Patent family member(s)			Publication date
US 6075821	A	13-06-2000	NONE			
WO 9963427	A	09-12-1999	US	6292539	B1	18-09-2001
			AU	4087599	A	20-12-1999
			EP	1093609	A1	25-04-2001
			WO	9963427	A1	09-12-1999
EP 0889615	A	07-01-1999	US	6252900	B1	26-06-2001
			EP	0889615	A2	07-01-1999
			JP	11088920	A	30-03-1999
			US	6088385	A	11-07-2000
			US	2001007574	A1	12-07-2001
			US	6442195	B1	27-08-2002
US 5128619	A	07-07-1992	EP	0391312	A2	10-10-1990
			JP	2292919	A	04-12-1990

Form PCT/ISA/210 (patent family annex) (July 1992)

Electronic Acknowledgement Receipt

EFS ID:	7782419
Application Number:	12779660
International Application Number:	
Confirmation Number:	8981
Title of Invention:	Multicarrier Modulation Messaging for Power Level Per Subchannel Information
First Named Inventor/Applicant Name:	David M. Krinsky
Customer Number:	62574
Filer:	Jason Vick/Debra Kesner
Filer Authorized By:	Jason Vick
Attorney Docket Number:	5550-2-CON2-1-1
Receipt Date:	09-JUN-2010
Filing Date:	13-MAY-2010
Time Stamp:	18:47:37
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1		IDS_01.pdf	693621 96b0beff7af3e5469d2a6372be8084de717116e4	yes	7

	Multipart Description/PDF files in .zip description				
	Document Description		**Start**	**End**	
	Transmittal Letter		1	3	
	Information Disclosure Statement (IDS) Filed (SB/08)		4	7	
Warnings:					
Information:					
2	Foreign Reference	EP0889615A2.pdf	1848555 931838aa76d186918d6307adbfea64690a80e368	no	34
Warnings:					
Information:					
3	Foreign Reference	GB2303032A.pdf	775828 971ccd21e6ca715331e9fd91a66779a4b3d7a218	no	24
Warnings:					
Information:					
4	Foreign Reference	JP_Hei6_1994-003956.pdf	1192136 798b375a7605b669bc529b15224f19aaa9624e95	no	15
Warnings:					
Information:					
5	Foreign Reference	JP10513622T.pdf	3431375 888b9862465b4615c95ce831cee3890f52d0c1ea	no	145
Warnings:					
Information:					
6	Foreign Reference	JP11261665A.pdf	935881 cfa6958f8b6b879b468fa4f2332329bb306cd9de	no	19
Warnings:					
Information:					
7	Foreign Reference	JP11317723A.pdf	471851 4faff8ed1775109955d76b796b73f4e16bb62b62	no	12
Warnings:					
Information:					
8	Foreign Reference	JP11508417T.pdf	1164326 a3796603ff4abfa9d2beb96f0ab15f9a5a117ff7	no	47
Warnings:					
Information:					

9	Foreign Reference	WO0064130A2.pdf	2380836 8a0213e66164e2b763e631da8dd091e1e12e61f7	no	74
Warnings:					
Information:					
10	Foreign Reference	WO8607223.pdf	2305273 3376921f387058b8b4d2f445ed88e8b477016982	no	53
Warnings:					
Information:					
11	Foreign Reference	WO9701900A1.pdf	1066593 32287592098a865152b537c1ac279cda6fdd099a	no	20
Warnings:					
Information:					
12	Foreign Reference	WO99020027.pdf	3477643 e650eb00032386da3e9b5646deca940758b888d8	no	71
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Information:					
13	Foreign Reference	WO9926375A2.pdf	1458669 94f3868d54ccceb0cb19877c8e7fb749b6637ff7	no	31
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Information:					
14	Foreign Reference	WO9963427A1.pdf	986542 d3d96d3f5ccc62ca2d9be876b4e36d6198fc3662	no	23
Warnings:					
Information:					
15	Foreign Reference	WO9967890A1.pdf	1092195 201646c9374aee2813049e34385b2b5638721d80	no	25
Warnings:					
Information:					
16	NPL Documents	Boets_Modeling_Aspect_of_Transmission_Line_Networks.pdf	399671 e5aabc3c165da136812ac990d82068488564dd13	no	6
Warnings:					
Information:					
17	NPL Documents	Cioffi_ADSL_Maintenance_with_DMT_T1E1_4.pdf	764298 a24c4bf9425f10d9789e1f0110fe4fc5429b31c4	no	14
Warnings:					
Information:					

18	NPL Documents	Lewis_Extending_Trouble_Ticket_System_to_Fault_Diagnostics.pdf	801049 16aff63faa90c450f6cb78fdb03c7a26a49d0499	no	8
Warnings:					
Information:					
19	NPL Documents	5550-2-PCT_Search_Report.pdf	156202 ce80d36e54318f42cd19132342bee0ba2645a706	no	4
Warnings:					
Information:					
20	NPL Documents	5550-2-PCT_Written_Opinion.pdf	241613 635fc12cf9d6917802d2146cce503dd896125fd7	no	2
Warnings:					
Information:					
21	NPL Documents	5550-2-PCT_IPER.pdf	85503 dfb67c85e9bdcc84d0df1b969e85eaecf8ee0d83	no	2
Warnings:					
Information:					
22	NPL Documents	5550-2-PCT-3_Search_Report.pdf	146869 884805bc2f978cbd8b79adb5d8e30c17bd636b5f	no	3
Warnings:					
Information:					
23	NPL Documents	5550-2-PAU_OA_4-2-04.pdf	56750 0d254394024b482c9a66a3be9b3129ecfed146fc	no	1
Warnings:					
Information:					
24	NPL Documents	5550-2-PAU_OA_8-6-04.pdf	89133 bbe57d04fd2ca68ddc5f6cee83798870a58fb996	no	2
Warnings:					
Information:					
25	NPL Documents	5550-2-PAU4_Examiners_First_Report_for_AU2004203321_11-16-06.pdf	131297 4ff0b79b6b7cf7cdcb00e51ab03715623ad52424	no	2
Warnings:					
Information:					
26	NPL Documents	5550-2-PAU4-DIV_OA_3-9-09.pdf	72889 2f41e077d06ead84bbb4e4b1ab9c984ffadeea72	no	2
Warnings:					
Information:					

27	NPL Documents	5550-2-PAU-4-DIV_NOA_7-9-09.pdf	64074 314f5bda4dc26bc9ae7148ce5c7b16d51d700128	no	2
Warnings:					
Information:					
28	NPL Documents	5550-2-PCA_OA_11-24-09.pdf	55806 5c821fd147f6f958641910c85b7056edf2845f44	no	2
Warnings:					
Information:					
29	NPL Documents	5550-2-PEP_OA_12-1-04.pdf	143351 c0d67e7101c9b24ab6ba46b14594788294b2b3fe	no	4
Warnings:					
Information:					
30	NPL Documents	5550-2-PEP_OA_9-14-05.pdf	52815 b7ea017eecfedb028e413830faf6689cdbcb0caf	no	2
Warnings:					
Information:					
31	NPL Documents	5550-2-PEP_NOA_5-15-06.pdf	1363474 b9ad07b60b6f0b36ae37596a08b3e30ea52ed522	no	25
Warnings:					
Information:					
32	NPL Documents	5550-2-PEP5_European_Search_Report_for_EP_06022008_1-8-07.pdf	696810 9ae058a5710f5b8d193a484700c79ac3032c01db	no	8
Warnings:					
Information:					
33	NPL Documents	5550-2-PEP5_OA_4-23-10.pdf	256735 fddbff1ef2c803a4dbd6bb178541cf1968032a84	no	9
Warnings:					
Information:					
34	NPL Documents	5550-2-PJP_OA_12-7-09.pdf	325581 9bb7afb456a2a1034a70d0efb7055792dbee6bfd	no	8
Warnings:					
Information:					
35	NPL Documents	5550-2-PKR_NOA_12-1-06.pdf	112458 f51253e094455b82ddcfda340ca66f0571d844f4	no	3
Warnings:					
Information:					

36	NPL Documents	5550-2-CON-2-1_OA_6-8-10.pdf	525546 85ca75fd2e078ff49fd99a3e45a5f41a337571fc	no	15
Warnings:					
Information:					
		Total Files Size (in bytes):	29823248		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

<u>New Applications Under 35 U.S.C. 111</u>
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

<u>National Stage of an International Application under 35 U.S.C. 371</u>
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

<u>New International Application Filed with the USPTO as a Receiving Office</u>
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In Re the Application of:	)	Group Art Unit: 2611
David M. Krinsky	))	Confirmation No.: 8981
Serial No.: 12/779,660	)	Examiner: Not yet assigned
Filed: May 13, 2010	))	
Atty. File No.: 5550-2-CON-2-1-1	)	INFORMATION DISCLOSURE STATEMENT
Entitled: "Multicarrier Modulation Messaging for	))	
Power Level Per Subchannel Information"	)	Electronically Submitted

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

The references cited on attached Form PTO-1449 are being called to the attention of the Examiner.

☒ Copies of the cited non-patent and/or foreign references are enclosed herewith.

☐ Copies of the cited U.S. patents and/or patent applications are enclosed herewith.

☒ Copies of the cited U.S. patents/patent application publications are not enclosed in accordance with 37 C.F.R. § 1.98(a).

☐ Copies of the cited references are not enclosed, in accordance with 37 C.F.R. § 1.98(d), because the references were cited by or submitted to the U.S. Patent and Trademark Office in prior application Serial No. ___________ filed ________________, which is relied upon for an earlier filing date under 35 U.S.C. § 120.

☒ To the best of applicants' belief, the pertinence of the foreign-language references are believed to be summarized in the attached English abstracts and in the figures, although applicants do not necessarily vouch for the accuracy of the translation.

☒ Examiner's attention is drawn to the following related applications:

Serial No. 09/755173 filed 01-08-2001, now U.S. Patent No. 6658052 (Attorney's Ref. No. 5550-2)

Serial No. 10/619691 filed 07-16-2003, now U.S. Patent No. 7570686 (Attorney's Ref. No. 5550-2-CON-2)

Serial No. 12/477742 filed 06-03-2009, now U.S. Patent Publication No. 2009/0238254 (Attorney's Ref. No. 5550-2-CON-2-1)

Serial No. 12/779708 filed 05-13-2010 (Attorney's Ref. No. 5550-2-CON-2-1-2)

☐ Other:____________________

Submission of the above information is not intended as an admission that any item is citable under the statutes or rules to support a rejection, that any item disclosed represents analogous art, or that those skilled in the art would refer to or recognize the pertinence of any reference without the benefit of hindsight, nor should an inference be drawn as to the pertinence of the references based on the order in which they are presented. Submission of this statement should not be taken as an indication that a search has been conducted, or that no better art exists.

It is respectfully requested that the cited information be expressly considered during the prosecution of this application and the references made of record therein.

FEES

☒	**37 CFR 1.97(b)**: No fee is believed due in connection with this submission, because the information disclosure statement submitted herewith is satisfied by one of the following conditions ("X" indicates satisfaction): ☒ Within three months of the filing date of a national application other than a continued prosecution application under 37 CFR 1.53(d), or ☐ Within three months of the date of entry into the national stage of an international application as set forth in 37 CFR 1.491 or ☐ Before the mailing date of a first Office Action on the merits, or ☐ Before the mailing of a first Office action after the filing of a request for continued examination under 37 CFR 1.114. Although no fee is believed due, if any fee is deemed due in connection with this submission, please charge such fee to Deposit Account 19-1970.
☐	**37 CFR 1.97(c)**: The information disclosure statement transmitted herewith is being filed after all the above conditions (37 CFR 1.97(b)), but before the mailing date of one of the following conditions: (1) a final action under 37 C.F.R. 1.113 or (2) a notice of allowance under 37 C.F.R. 1.311, or (3) an action that otherwise closes prosecution in the application. This Information Disclosure Statement is accompanied by: ☐ A Certification (below) as specified by 37 C.F.R. 1.97(e). Although no fee is believed due, if any fee is deemed due in connection with this submission, please charge such fee to Deposit Account 19-1970. OR ☐ Please charge Deposit Account 19-1970 in the amount of $180.00 for the fee set forth in 37 C.F.R. 1.17(p) for submission of an information disclosure statement. Please credit any overpayment or charge any underpayment to Deposit Account 19-1970.
☐	**37 CFR 1.97(d)**: This Information Disclosure Statement is being submitted after the period specified in 37 CFR 1.97(c). ☐ This information Disclosure Statement includes a Certification (below) as specified by 37 C.F.R. 1.97(e) AND ☐ Applicants hereby requests consideration of the reference(s) disclosed herein. Please charge Deposit Account 19-1970 in the amount of $180.00 under 37 C.F.R. 1.17(p). Please credit any overpayment or charge any underpayment to Deposit Account 19-1970. Election to pay the fee should not be taken as an indication that applicant(s) cannot execute a certification.

Certification (37 C.F.R. 1.97(e))
(Applicable only if checked)

☐ The undersigned certifies that:

☐ Each item of information contained in this information disclosure statement was first cited in any communication from a foreign patent office in a counterpart foreign application not more than three months prior to the filing of this statement. 37 C.F.R. 1.97(e)(1).

☐ A copy of the communication from the foreign patent office is enclosed.

OR

☐ No item of information contained in this information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the undersigned after making reasonable inquiry, no item of information contained in this Information Disclosure Statement was known to any individual designated in 37 C.F.R. 1.56(c) more than three months prior to the filing of this statement. 37 C.F.R. 1.97(e)(2).

Respectfully submitted,

SHERIDAN ROSS P.C.

By: ______________________

Jason H. Vick
Registration No. 45,285
1560 Broadway, Suite 1200
Denver, Colorado 80202-5141
(303) 863-9700

Date: 7 Jan '10

UNITED STATES PATENT AND TRADEMARK OFFICE

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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
12/779,660	05/13/2010	2611	1750	5550-2-CON2-1-1	12	6

CONFIRMATION NO. 8981

62574
Jason H. Vick
Sheridan Ross, PC
Suite # 1200
1560 Broadway
Denver, CO 80202

FILING RECEIPT

OC000000041799505

Date Mailed: 06/01/2010

Receipt is acknowledged of this non-provisional patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF APPLICANT, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection. Please verify the accuracy of the data presented on this receipt. **If an error is noted on this Filing Receipt, please submit a written request for a Filing Receipt Correction. Please provide a copy of this Filing Receipt with the changes noted thereon. If you received a "Notice to File Missing Parts" for this application, please submit any corrections to this Filing Receipt with your reply to the Notice. When the USPTO processes the reply to the Notice, the USPTO will generate another Filing Receipt incorporating the requested corrections**

Applicant(s)

David M. Krinsky, Acton, MA;
Robert Edmund Pizzano JR., Stoneham, MA;

Assignment For Published Patent Application

AWARE, INC., Bedford, MA

Power of Attorney: None

Domestic Priority data as claimed by applicant

This application is a CON of 12/477,742 06/03/2009
which is a CON of 10/619,691 07/16/2003 PAT 7,570,686
which is a DIV of 09/755,173 01/08/2001 PAT 6,658,052
which claims benefit of 60/224,308 08/10/2000
and claims benefit of 60/174,865 01/07/2000

Foreign Applications

If Required, Foreign Filing License Granted: 05/25/2010

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 12/779,660**

Projected Publication Date: 09/09/2010

Non-Publication Request: No

Early Publication Request: No

Title

Multicarrier Modulation Messaging for Power Level Per Subchannel Information

Preliminary Class

375

PROTECTING YOUR INVENTION OUTSIDE THE UNITED STATES

Since the rights granted by a U.S. patent extend only throughout the territory of the United States and have no effect in a foreign country, an inventor who wishes patent protection in another country must apply for a patent in a specific country or in regional patent offices. Applicants may wish to consider the filing of an international application under the Patent Cooperation Treaty (PCT). An international (PCT) application generally has the same effect as a regular national patent application in each PCT-member country. The PCT process **simplifies** the filing of patent applications on the same invention in member countries, but **does not result** in a grant of "an international patent" and does not eliminate the need of applicants to file additional documents and fees in countries where patent protection is desired.

Almost every country has its own patent law, and a person desiring a patent in a particular country must make an application for patent in that country in accordance with its particular laws. Since the laws of many countries differ in various respects from the patent law of the United States, applicants are advised to seek guidance from specific foreign countries to ensure that patent rights are not lost prematurely.

Applicants also are advised that in the case of inventions made in the United States, the Director of the USPTO must issue a license before applicants can apply for a patent in a foreign country. The filing of a U.S. patent application serves as a request for a foreign filing license. The application's filing receipt contains further information and guidance as to the status of applicant's license for foreign filing.

Applicants may wish to consult the USPTO booklet, "General Information Concerning Patents" (specifically, the section entitled "Treaties and Foreign Patents") for more information on timeframes and deadlines for filing foreign patent applications. The guide is available either by contacting the USPTO Contact Center at 800-786-9199, or it can be viewed on the USPTO website at http://www.uspto.gov/web/offices/pac/doc/general/index.html.

For information on preventing theft of your intellectual property (patents, trademarks and copyrights), you may wish to consult the U.S. Government website, http://www.stopfakes.gov. Part of a Department of Commerce initiative, this website includes self-help "toolkits" giving innovators guidance on how to protect intellectual property in specific countries such as China, Korea and Mexico. For questions regarding patent enforcement issues, applicants may call the U.S. Government hotline at 1-866-999-HALT (1-866-999-4158).

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Title 35, United States Code, Section 184

Title 37, Code of Federal Regulations, 5.11 & 5.15

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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
12/779,660	05/13/2010	David M. Krinsky	5550-2-CON2-1-1

CONFIRMATION NO. 8981

IMPROPER CPOA LETTER

OC000000041799511

62574
Jason H. Vick
Sheridan Ross, PC
Suite # 1200
1560 Broadway
Denver, CO 80202

Date Mailed: 06/01/2010

NOTICE REGARDING POWER OF ATTORNEY

This is in response to the Power of Attorney filed 05/13/2010. The Power of Attorney in this application is not accepted for the reason(s) listed below:

- The Power of Attorney you provided did not comply with the new Power of Attorney rules that became effective on June 25, 2004. See 37 CFR 1.32.

/asahle/

Office of Data Management, Application Assistance Unit (571) 272-4000, or (571) 272-4200, or 1-888-786-0101

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In Re the Application of: David M. Krinsky	))	Group Art Unit:
Application No.:	))	Examiner:
Filed: Herewith	))	Confirmation No.:
Atty. File No.: 5550-2-CON2-1-1	))	

For: MULTICARRIER MODULATION MESSAGING FOR POWER LEVEL PER SUBCHANNEL INFORMATION (As Amended)

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313

PRELIMINARY AMENDMENT

Dear Sir:

Prior to the initial review of the above-identified patent application by the Examiner, please enter the following Preliminary Amendment. Although Applicants do not believe that any fees are due based upon the filing of this Preliminary Amendment, please charge any such fees to Deposit Account 19-1970.

Please amend the above-identified patent application as follows:

Amendments to the Specification begin on page 2 of this paper.

Amendments to the Claims are shown in the listing of claims which begin on page 3 of this paper.

Remarks begin on page 6 of this paper.

AMENDMENTS TO THE SPECIFICATION

Please change the title to read as follows:

MULTICARRIER MODULATION MESSAGING FOR POWER LEVEL PER SUBCHANNEL INFORMATION

Please insert the following paragraph as the first paragraph beneath the title:

Related Application Data

This application is a continuation of U.S. Application No. 12/477,742, filed June 3, 2009, which is a continuation of U.S. Application No. 10/619,691, filed July 16, 2003, now U.S. Patent No. 7,570,686, which is a divisional of U.S. Application No. 09/755,173, filed January 8, 2001, now U.S. Patent No. 6,658,052, which claims the benefit of and priority under 35 U.S.C. §119(e) to U.S. Provisional Application No. 60/224,308, filed August 10, 2000 entitled "Characterization of transmission lines using broadband signals in a multi-carrier DSL system," and U.S. Provisional Application No. 60/174,865, filed January 7, 2000 entitled "Multicarrier Modulation System with Remote Diagnostic Transmission Mode", each of which are incorporated herein by reference in their entirety.

AMENDMENTS TO THE CLAIMS:

This listing of claims will replace all prior versions, and listings, of claims in the application.

Listing of Claims:

1. – 43. (Cancelled)

44. (New) A transceiver capable of transmitting test information over a communication channel using multicarrier modulation comprising:

a transmitter portion capable of transmitting a message, wherein the message comprises one or more data variables that represent the test information, wherein bits in the message are modulated onto DMT symbols using Quadrature Amplitude Modulation (QAM) with more than 1 bit per subchannel and wherein at least one data variable of the one or more data variables comprises an array representing power level per subchannel information.

45. (New) The transceiver of claim 44, wherein the power level per subchannel information is based on a Reverb signal.

46. (New) A transceiver capable of receiving test information over a communication channel using multicarrier modulation comprising:

a receiver portion capable of receiving a message, wherein the message comprises one or more data variables that represent the test information, wherein bits in the message were modulated onto DMT symbols using Quadrature Amplitude Modulation (QAM) with more than 1 bit per subchannel and wherein at least one data variable of the one or more data variables comprises an array representing power level per subchannel information.

47. (New) The transceiver of claim 46, wherein the power level per subchannel information is based on a Reverb signal.

48. (New) In a transceiver capable of transmitting test information over a communication channel using multicarrier modulation, a method comprising:

transmitting a message, wherein the message comprises one or more data variables that represent the test information, wherein bits in the message are modulated onto DMT symbols using Quadrature Amplitude Modulation (QAM) with more than 1 bit per subchannel and wherein at least one data variable of the one or more data variables comprises an array representing power level per subchannel information.

49. (New) The method of claim 48, wherein the power level per subchannel information is based on a Reverb signal.

50. (New) In a transceiver capable of receiving test information over a communication channel using multicarrier modulation, a method comprising:

receiving a message, wherein the message comprises one or more data variables that represent the test information, wherein bits in the message were modulated onto DMT symbols using Quadrature Amplitude Modulation (QAM) with more than 1 bit per subchannel and wherein at least one data variable of the one or more data variables comprises an array representing power level per subchannel information.

51. (New) The method of claim 50, wherein the power level per subchannel information is based on a Reverb signal.

52. (New) A non-transitory computer-readable information storage media having stored thereon instructions that, if executed, cause a transceiver to perform a method comprising:

transmitting a message, wherein the message comprises one or more data variables that represent the test information, wherein bits in the message are modulated onto DMT symbols using Quadrature Amplitude Modulation (QAM) with more than 1 bit per subchannel and wherein at least one data variable of the one or more data variables comprises an array representing power level per subchannel information.

53. (New) The media of claim 52, wherein the power level per subchannel information is based on a Reverb signal.

54. (New) A non-transitory computer-readable information storage media having stored thereon instructions that, if executed, cause a transceiver to perform a method comprising:

receiving a message, wherein the message comprises one or more data variables that represent the test information, wherein bits in the message were modulated onto DMT symbols using Quadrature Amplitude Modulation (QAM) with more than 1 bit per subchannel and wherein at least one data variable of the one or more data variables comprises an array representing power level per subchannel information.

55. (New) The media of claim 54, wherein the power level per subchannel information is based on a Reverb signal.

REMARKS/ARGUMENTS

By this amendment, claim 1-43 have been cancelled without prejudice or disclaimer in favor of the newly presented claims 44-55 that are directed toward more specific aspects of the invention.

Applicant requests examination on the merits.

Applicant believes that the pending claims are in condition for allowance and such disposition is respectfully requested. In the event that a telephone conversation would further prosecution and/or expedite allowance, the Examiner is invited to contact the undersigned.

The Commissioner is hereby authorized to charge to Deposit Account No. 19-1970 any fees under 37 C.F.R. §§ 1.16 and 1.17 that may be required by this paper and to credit any overpayment to that Account. If any extension of time is required in connection with the filing of this paper and has not been separately requested, such extension is hereby Petitioned.

Respectfully submitted,

SHERIDAN ROSS P.C.

By: ______________________

Jason H. Vick
Registration No. 45,285
1560 Broadway, Suite 1200
Denver, Colorado 80202-5141
(303) 863-9700

Date: 13 Mar '16

SYSTEMS AND METHODS FOR ESTABLISHING A DIAGNOSTIC TRANSMISSION MODE AND COMMUNICATING OVER THE SAME

Field of the Invention

This invention relates to test and diagnostic information. In particular, this invention relates to a robust system and method for communicating diagnostic information.

Background of the Invention

The exchange of diagnostic and test information between transceivers in a telecommunications environment is an important part of a telecommunications, such as an ADSL, deployment. In cases where the transceiver connection is not performing as expected, for example, where the data rate is low, where there are many bit errors, or the like, it is necessary to collect diagnostic and test information from the remote transceiver. This is performed by dispatching a technician to the remote site, e.g., a truck roll, which is time consuming and expensive.

In DSL technology, communications over a local subscriber loop between a central office and a subscriber premises is accomplished by modulating the data to be transmitted onto a multiplicity of discrete frequency carriers which are summed together and then transmitted over the subscriber loop. Individually, the carriers form discrete, non-overlapping communication subchannels of limited bandwidth. Collectively, the carriers form what is effectively a broadband communications channel. At the receiver end, the carriers are demodulated and the data recovered.

DSL systems experience disturbances from other data services on adjacent phone lines, such as, for example, ADSL, HDSL, ISDN, T1, or the like. These disturbances may commence after the subject ADSL service is already initiated and, since DSL for internet access is envisioned as an always-on service, the effect of these disturbances must be ameliorated by the subject ADSL transceiver.

SUMMARY OF THE INVENTION

The systems and methods of this invention are directed toward reliably exchanging diagnostic and test information between transceivers over a digital subscriber line in the presence of voice communications and/or other disturbances. For simplicity of reference, the systems and methods of the invention will hereafter refer to the transceivers generically as modems. One such modem is typically located at a customer premises such as a home or business and is "downstream" from a central office with which it communicates. The other modem is typically located at the central office and is "upstream" from the customer premises. Consistent with industry practice, the modems are often referred to as "ATU-R" ("ADSL transceiver unit, remote," i.e., located at the customer premises) and "ATU-C" ("ADSL transceiver unit, central office" i.e., located at the central office). Each modem includes a transmitter section for transmitting data and a receiver section for receiving data, and is of the discrete multitone type, i.e., the modem transmits data over a multiplicity of subchannels of limited bandwidth. Typically, the upstream or ATU-C modem transmits data to the downstream or ATU-R modem over a first set of subchannels, which are usually the higher-frequency subchannels, and receives data from the downstream or ATU-R modem over a second, usually smaller, set of subchannels, commonly the lower-frequency subchannels. By establishing a diagnostic link mode between the two modems, the systems and methods of this invention are able to exchange diagnostic and test information in a simple and robust manner.

In the diagnostic link mode, the diagnostic and test information is communicated using a signaling mechanism that has a very high immunity to noise and/or other disturbances and can therefore operate effectively even in the case where the modems could not actually establish an acceptable connection in their normal operational mode.

For example, if the ATU-C and/or ATU-R modem fail to complete an initialization sequence, and are thus unable to enter a normal steady state communications mode, where the diagnostic and test information would normally be exchanged, the modems according to the systems and methods of this invention enter a robust diagnostic link mode. Alternatively, the diagnostic link mode can be entered automatically or manually, for example, at the direction of a user. In the robust diagnostic link mode, the modems exchange the diagnostic and test information that is, for example, used by a technician to determine the cause of a failure without the technician having to physically visit, i.e., a truckroll to, the remote site to collect data.

The diagnostic and test information can include, for example, but is not limited to, signal to noise ratio information, equalizer information, programmable gain setting information, bit allocation information, transmitted and received power information, margin information, status and rate information, telephone line condition information, such as the length of the line, the number and location of bridged taps, a wire gauge, or the like, or any other known or later developed diagnostic or test information that may be appropriate for the particular communications environment. For example, the exchanged diagnostic and test information can be directed toward specific limitations of the modems, to information relating to the modem installation and deployment environment, or to other diagnostic and test information that can, for example, be determined as needed which may aid in evaluating the cause of a specific failure or problem. Alternatively, the diagnostic and test information can include the loop length and bridged tap length estimations as discussed in copending Attorney Docket No. 081513-000003, filed herewith and incorporated herein by reference in its entirety.

For example, an exemplary embodiment of the invention illustrates the use of the diagnostic link mode in the communication of diagnostic information from the remote terminal (RT) transceiver, e.g., ATU-R, to the central office (CO) transceiver, e.g., ATU-C. Transmission of information from the remote terminal to the central office is important since a typical ADSL service provider is located in the central office and would therefore benefit from the ability to determine problems at the remote terminal without a truckroll. However, it is to be appreciated, that the systems and the methods of this invention will work equally well in communications from the central office to the remote terminal.

These and other features and advantages of this invention are described in or are apparent from the following detailed description of the embodiments.

BRIEF DESCRIPTION OF THE DRAWINGS

The embodiments of the invention will be described in detail, with reference to the following figures wherein:

Fig. 1 is a functional block diagram illustrating an exemplary communications system according to this invention; and

Fig. 2 is a flowchart outlining an exemplary method for communicating diagnostic and test information according to this invention.

DETAILED DESCRIPTION OF THE INVENTION

For ease of illustration the following description will be described in relation to the CO receiving diagnostic and test information from the RT. In the exemplary embodiment, the systems and methods of this invention complete a portion of the normal modem initialization before entering into the diagnostic link mode. The systems and methods of this invention can enter the diagnostic link mode manually, for example, at the direction of a technician or a user after completing a portion of initialization. Alternatively, the systems and methods of this invention can enter the diagnostic link mode automatically based on, for example, a bit rate failure, a forward error correction or a CRC error during showtime, e.g., the normal steady state transmission mode, or the like. The transition into the diagnostic link mode is accomplished by transmitting a message from the CO modem to the RT modem indicating that the modems are to enter into the diagnostic link mode, as opposed to transitioning into the normal steady state data transmission mode. Alternatively, the transition into the diagnostic link mode is accomplished by transmitting a message from the RT modem to the CO modem indicating that the modems are to enter into the diagnostic link mode as opposed to transitioning into the normal steady state data transmission mode. For example, the transition signal uses an ADSL state transition to transition from a standard ADSL state to a diagnostic link mode state.

In the diagnostic link mode, the RT modem sends diagnostic and test information in the form of a collection of information bits to the CO modem that are, for example, modulated by using one bit per DTM symbol modulation, as is used in the C-Rates1 message in the ITU and ANSI ADSL standards, where the symbol may or may not include a cyclic prefix. Other exemplary modulation techniques include Differential Phase Shift Keying (DPSK) on a subset or all the carriers, as specified in, for example, ITU standard G.994.1, higher order QAM modulation (>1 bit per carrier), or the like.

In the one bit per DMT symbol modulation message encoding scheme, a bit with value 0 is mapped to the REVERB1 signal and a bit with a value of 1 mapped to a SEGUE1 signal. The REVERB1 and SEGUE1 signals are defined in the ITU and ANSI ADSL standards. The REVERB1 signal is generated by modulating all of the carriers in the multicarrier system with a known pseudo-random sequence thus generating a wideband modulated signal. The SEGUE1 signal is generated from a carrier by 180 degree phase reversal of the REVERB1 signal. Since both signals are wideband and known in advance, the receiver can easily detect the REVERB1 and SEGUE1 signal using a simple matched filter in the presence of large amounts of noise and other disturbances.

Exemplary Message Variables
Data Sent in the Diag Link
Train Type
ADSL Standard
Chip Type
Vendor ID
Code Version
Average Reverb Received Signal
Programmable gain amplifier (PGA) Gain – Training
Programmable gain amplifier PGA Gain – Showtime
Filter Present during Idle Channel Calculation
Average Idle Channel Noise
Signal to Noise during Training
Signal to Noise during Showtime
Bits and Gains
Data Rate
Framing Mode
Margin
Reed-Solomon Coding Gain
QAM Usage
Frequency Domain Equalizer (FDQ) Coefficients
Gain Scale
Time domain equalizer (TDQ) Coefficients
Digital Echo Canceller (DEC) Coefficients

Table 1

Table 1 shows an example of a data message that can be sent by the RT to the CO during the diagnostic link mode. In this example, the RT modem sends 23 different data variables to the CO. Each data variable contains different items of diagnostic and test information that are used to analyze the condition of the link. The variables may contain more than one item of data. For example, the *Average Reverb Signal* contains the power levels per tone, up to, for example, 256 entries, detected during the ADSL Reverb signal. Conversely, the *PGA Gain – Training* is a single entry, denoting the gain in dB at the receiver during the ADSL training.

Many variables that represent the type of diagnostic and test information that are used to analyze the condition of the link are sent from the RT modem to the CO modem. These variables can be, for example, arrays with different lengths depending on, for example, information in the initiate diagnostic mode message. The systems and methods of this invention can be tailored to contain many different diagnostic and test information variables.

Thus, the system is fully configurable, allowing subsets of data to be sent and additional data variables to be added in the future. Therefore, the message length can be increased or decreased, and diagnostic and test information customized, to support more or less variables as, for example, hardware, the environment and/or the telecommunications equipment dictates.

Therefore, it is to be appreciated, that in general the variables transmitted from the modem being tested to the receiving modem can be any combination of variables which allow for transmission of test and/or diagnostic information.

Fig. 1 illustrates an exemplary embodiment of the additional modem components associated with the diagnostic link mode. In particular, the diagnostic link system 100 comprises a central office modem 200 and a remote terminal modem 300. The central office modem 200 comprises, in addition to the standard ATU-C components, a CRC checker 210, a diagnostic device 220, and a diagnostic information monitoring device 230. The remote terminal modem 300 comprises, in addition to the standard components associated with an ATU-R, a message determination device 310, a power control device 320, a diagnostic device 330 and a diagnostic information storage device 340. The central office modem 200 and the remote terminal model 300 are also connected, via link 5, to a splitter 10 for a phone switch 20, and a splitter 30 for a phone 40. Alternatively, the ATU-R can operate without a splitter, e.g., splitterless, as specified in ITU standard G.992.2 (G.lite) or with an in-line filter in series with the phone 40. In addition, the remote terminal modem 300, can also be connected to, for example, one or more user terminals 60. Additionally, the central office modem 200 can be connected to one or more distributed networks 50, via link 5, which may or may not also be connected to one or more other distributed networks.

While the exemplary embodiment illustrated in Fig. 1 shows the diagnostic link system 100 for an embodiment in which the remote terminal modem 300 is communicating test and diagnostic information to the central office 200, it is to be appreciated that the various components of the diagnostic link system can be rearranged such that the diagnostic and test information can be forwarded from the central office 200 to the remote terminal modem 300, or, alternatively, such that both modems can send and receive diagnostic and/or test information. Furthermore, it is to be appreciated, that the components of the diagnostic link system 100 can be located at various locations within a distributed network, such as the - POTS network, or other comparable telecommunications network. Thus, it should be appreciated that the components of the diagnostic link system 100 can be combined into one device for respectively transmitting, receiving, or transmitting and receiving diagnostic and/or test information. As will be appreciated from the following description, and for reasons of computational efficiency, the components of the diagnostic link system 100 can be arranged at any location within a telecommunications network and/or modem without affecting the operation of the system.

The links 5 can be a wired or wireless link or any other known or later developed element(s) that is capable of supplying and communicating electronic data to and from the connected elements. Additionally, the user terminal 60 can be, for example, a personal computer or other device allowing a user to interface with and communicate over a modem, such as a DSL modem. Furthermore, the systems and method of this invention will work equally well with splitterless and low-pass mulitcarrier modem technologies.

In operation, the remote terminal 300, commences its normal initialization sequence. The diagnostic device 330 monitors the initialization sequence for a failure. If there is a failure, the diagnostic device 330 initiates the diagnostic link mode. Alternatively, a user or, for example, a technician at the CO, can specify that the remote terminal 300 enter into the diagnostic link mode after completing a portion of an initialization. Alternatively still, the diagnostic device 330 can monitor the normal steady state data transmission of the remote terminal, and upon, for example, an error threshold being exceeded, the diagnostic device 330 will initiate the diagnostic link mode.

Upon initialization of the diagnostic link mode, the diagnostic device 330, in cooperation with the remote terminal 300 will transmit an initiate diagnostic link mode message from the remote terminal to the central office 200 (RT to CO). Alternatively, the central office modem 200 can transmit an initiate diagnostic link mode message to the remote terminal modem 300. If the initiate diagnostic link mode message is received by the central office 200, the diagnostic device 330, in cooperation with the message determination device 310, determines a diagnostic link message to be forwarded to the central office 200. For example, the diagnostic link message can include test information that has been assembled during, for example, the normal ADSL initialization procedure. The diagnostic and/or test information can include, but is not limited to, the version number of the diagnostic link mode, the length of the diagnostic and/or test information, the communications standard, such as the ADSL standard, the chipset type, the vendor identifications, the ATU version number, the time domain received reverb signal, the frequency domain reverb signal, the amplifier settings, the CO transmitter power spectral density, the frequency domain received idle channel, the signal to noise ratio, the bits and gains and the upstream and downstream transmission rates, or the like.

If the initiate diagnostic link mode message is not received by the central office 200, the initiate diagnostic link mode message can, for example, be re-transmitted a predetermined number of iterations until a determination is made that it is not possible to establish a connection.

Assuming the initiate diagnostic link mode message is received, then, for a predetermined number of iterations, the diagnostic device 330, in cooperation with the remote terminal modem 300 and the diagnostic information storage device 340, transmits the diagnostic link message with a cyclic redundancy check (CRC) to the central office modem 200. However, it is to be appreciated that in general, any error detection scheme, such as bit error detection, can be used without affecting the operation of the system. The central office 200, in cooperation with the CRC checker 210, determines if the CRC is correct. If the CRC is correct, the diagnostic information stored in the diagnostic information storage device 340 has been, with the cooperation of the diagnostic device 330, and the remote terminal modem 300, forwarded to the central office 200 successfully.

If, for example, the CRC checker 210 is unable to determine the correct CRC, the diagnostic device 330, in cooperation with power control device 320, increases the transmission power of the remote terminal 300 and repeats the transmission of the diagnostic link message from the remote terminal 300 to the central office 200. This process continues until the correct CRC is determined by the CRC checker 210.

The maximum power level used for transmission of the diagnostic link message can be specified by, for example, the user or the ADSL service operator. If the CRC checker 210 does not determine a correct CRC at the maximum power level and the diagnostic link mode can not be initiated then other methods for determining diagnostic information are utilized, such as dispatching a technician to the remote site, or the like.

Alternatively, the remote terminal 300, with or without an increase in the power level, can transmit the diagnostic link message several times, for example, 4 times. By transmitting the diagnostic link message several times, the CO modem 200 can use, for example, a diversity combining scheme to improve the probability of obtaining a correct CRC from the received diagnostic link message(s).

Alternatively, as previously discussed, the central office 200 comprises a diagnostic information monitoring device 230. The remote terminal 300 can also include a diagnostic information monitoring device. One or more of these diagnostic information monitoring devices can monitor the normal steady state data transmission between the remote terminal 300 and the central office 200. Upon, for example, the normal steady state data transmission exceeded a predetermined error threshold, the diagnostic information monitoring device can initiate the diagnostic link mode with the cooperation of the diagnostic device 300 and/or the diagnostic device 220.

Fig. 2 illustrates an exemplary method for entering a diagnostic link mode in accordance with this invention. In particular, control begins in step S100 and continues to step S110. In step S110, the initialization sequence is commenced. Next, in step S120, if an initialization failure is detected, control continues to step S170. Otherwise, control jumps to step S130. In step S130, a determination is made whether the diagnostic link mode has been selected. If the diagnostic link mode has been selected, control continues to step S170, otherwise, control jumps to step S140.

In step S170, the initiate diagnostic link mode message is transmitted from, for example, the remote terminal to the central office. Next, in step S180, a determination is made whether the initiate diagnostic mode message has been received by the CO. If the initiate diagnostic mode message has been received by the CO, control jumps to step S200. Otherwise, control continues to step S190. In step S190, a determination is made whether to re-transmit the initiate diagnostic mode message, for example, based on whether a predetermined number of iterations have already been completed. If the initiate diagnostic mode message is to be re-transmitted, control continues back to step S170. Otherwise, control jumps to step S160.

In step S200, the diagnostic link message is determined, for example, by assembling test and diagnostic information about one or more of the local loop, the modem itself, the telephone network at the remote terminal, or the like. Next, in step S210, for a predetermined number of iterations, steps S220-S240 are completed. In particular, in step S220 a diagnostic link message comprising a CRC is transmitted to, for example, the CO. Next, in step S230, the CRC is determined. Then, in step S240, a determination is made whether the CRC is correct. If the CRC is correct, the test and/or diagnostic information has been successfully communicated and control continues to step S160.

Otherwise, if step S210 has completed the predetermined number of iterations, control continues to step S250. In step S250, the transmission power is increased and control continues back to step S210. Alternatively, as previously discussed, the diagnostic link message may be transmitted a predetermined number of times, with our without a change in the transmission power.

In step S140, the normal steady state data transmission is entered into between two modems, such as the remote terminal and the cental office modems. Next, in step S150, a determination is made whether an error threshold during the normal steady state data transmission has been exceeded. If the error threshold has been exceeded, control continues to step S170. Otherwise, control jumps to step S160. In step S160, the control sequence ends.

As shown in Fig. 1, the diagnostic link mode system can be implemented either on a single program general purpose computer, a modem, such as a DSL modem, or a separate program general purpose computer having a communications device. However, the diagnostic link system can also be implemented on a special purpose computer, a programmed microprocessor or microcontroller and peripheral integrated circuit element, an ASIC or other integrated circuit, a digital signal processor, a hardwired electronic or logic circuit such as a discrete element circuit, a programmed logic device such as a PLD, PLA, FPGA, PAL, or the like, and associated communications equipment. In general, any device capable of implementing a finite state machine that is capable of implementing the flowchart illustrated in Fig. 2 can be used to implement a diagnostic link system according to this invention.

Furthermore, the disclosed method may be readily implemented in software using object or object-oriented software development environments that provide portable source code that can be used on a variety of computer, workstation, or modem hardware platforms. Alternatively, the disclosed diagnostic link system may be implemented partially or fully in hardware using standard logic circuits or a VLSI design. Other software or hardware can be used to implement the systems in accordance with this invention depending on the speed and/or efficiency requirements of the systems, the particular function, and a particular software or hardware systems or microprocessor or microcomputer systems being utilized. The diagnostic link system and methods illustrated herein however, can be readily implemented in hardware and/or software using any known or later developed systems or structures, devices and/or software by those of ordinary skill in the applicable art from the functional description provided herein and with a general basic knowledge of the computer and telecommunications arts.

Moreover, the disclosed methods can be readily implemented as software executed on a programmed general purpose computer, a special purpose computer, a microprocessor, or the like. In these instances, the methods and systems of this invention can be implemented as a program embedded on a modem, such a DSL modem, as a resource residing on a personal computer, as a routine embedded in a dedicated diagnostic link system, a central office, or the like. The diagnostic link system can also be implemented by physically incorporating the system and method into a software and/or hardware system, such as a hardware and software systems of a modem, a general purpose computer, an ADSL line testing device, or the like.

It is, therefore, apparent that there is provided in accordance with the present invention, systems and methods for transmitting a diagnostic link message. While this invention has been described in conjunction with a number of embodiments, it is evident that many alternatives, modifications and variations would be or are apparent to those of ordinary skill in the applicable arts. Accordingly, applicants intend to embrace all such alternatives, modifications, equivalents and variations that are within the spirit and the scope of this invention.

What is Claimed is:

1. A diagnostic link system for communicating data between modems using multicarrier modulation comprising:

an initiate diagnostic mode trigger that instructs a transmitting modem to forward an initiate diagnostic mode message to a receiving modem;

a message determination device that determines a diagnostic link message; and

a receiving modem diagnostic device that receives the diagnostic link message and determines the accuracy of the diagnostic link message.

2. The system of claim 1, further comprising a power control device that increases a transmission power of the diagnostic link message if the received diagnostic link message is inaccurate.

3. The system of claim 1, wherein the diagnostic link message is re-transmitted a predetermined number of times.

4. The system of claim 1, wherein the diagnostic link message comprises at least one of test and diagnostic information.

5. The system of claim 4, wherein the diagnostic link message comprises at least one of a version number of a diagnostic link mode, a length of the diagnostic information, a communications standard, a chipset type, one or more vendor identifications, an ATU version number, a time domain received reverb signal, a frequency domain reverb signal, an amplifier setting, a CO transmitter power spectral density, a frequency domain received idle channel, a signal to noise ratio, bits and gain information, and upstream and downstream transmission rates.

6. The system of claim 1, wherein the accuracy is determined based on at least one of an error detecting scheme, a bit error detection and a cyclic redundancy check.

7. The system of claim 1, wherein the trigger is based on at least one of an initialization failure, a bit rate failure, a CRC error in an initialization message, a CRC error during a normal steady state transmission mode, a forward error correction error, a user request, a central office modem request and a remote terminal modem request.

8. The system of claim 1, wherein the transmitting modem completes a portion of a modem initialization sequence before forwarding the initiate diagnostic mode message.

9. The system of claim 1, wherein the transmitting modem is at least one of a central office modem and a remote terminal modem.

10. The system of claim 1, wherein the receiving modem is at least one of a central office modem and a remote terminal modem.

11. A method for communicating data between modems using multicarrier modulation comprising:

instructing a transmitting modem to forward an initiate diagnostic mode message to a receiving modem;

determining a diagnostic link message;

transmitting the diagnostic link message; and

determining the accuracy of the transmitted diagnostic link message.

12. The method of claim 11, further comprising increasing a transmission power of the diagnostic link message if a received diagnostic link message is inaccurate.

13. The method of claim 11, further comprising transmitting the diagnostic link message a predetermined number of times.

14. The method of claim 11, wherein the diagnostic link message comprises at least one of test and diagnostic information.

15. The method of claim 14, wherein the diagnostic link message comprises at least one of a version number of a diagnostic link mode, a length of the diagnostic information, a communications standard, a chipset type, one or more vendor identifications, an ATU version number, a time domain received reverb signal, a frequency domain reverb signal, an amplifier setting, a CO transmitter power spectral density, a frequency domain received idle channel, a signal to noise ratio, bits and gain information, and upstream and downstream transmission rates.

16. The method of claim 11, wherein the accuracy is determined based on at least one of an error detecting scheme, a bit error detection and a cyclic redundancy check.

17. The method of claim 11, wherein the initiate diagnostic mode message is based on at least one of an initialization failure, a bit rate failure, a CRC error in an initialization message, a CRC error during the normal steady state transmission mode, a forward error correction error, a user request, a central office modem request and a remote terminal modem request.

18. The method of claim 11, further comprising completing a portion of a modem initialization sequence before forwarding the initiate diagnostic mode message.

19. The method of claim 11, wherein a transmitting modem is at least one of a central office modem and a remote terminal modem.

20. The method of claim 11, wherein a receiving modem is at least one of a central office modem and a remote terminal modem.

21. A method for communicating data between modems using multicarrier modulation comprising:

receiving an initiate diagnostic mode message;

determining a diagnostic link message;

transmitting the diagnostic link message; and

at least one of increasing a transmission power of the diagnostic link message if the received diagnostic link message is inaccurate and re-transmitting the diagnostic link message a predetermined number of times.

22. The method of claim 21, wherein the diagnostic link message comprises at least one of test and diagnostic information.

23. The method of claim 22, wherein the diagnostic link message comprises at least one of a version number of a diagnostic link mode, a length of the diagnostic information, a communications standard, a chipset type, one or more vendor identifications, an ATU version number, a time domain received reverb signal, a frequency domain reverb signal, an amplifier setting, a CO transmitter power spectral density, a frequency domain received idle channel, a signal to noise ratio, bits and gain information, and upstream and downstream transmission rates.

24. The method of claim 21, wherein the accuracy is determined based on at least one of an error detecting scheme, a bit error detection and a cyclic redundancy check.

25. The method of claim 21, wherein the initiate diagnostic mode message is based on at least one of an initialization failure, a bit rate failure, a CRC error in an initialization message, a CRC error during the normal steady state transmission mode, a forward error correction error, a user request, a central office modem request and a remote terminal modem request.

26. The method of claim 21, further comprising completing a portion of a modem initialization sequence before forwarding the initiate diagnostic mode message.

27. The method of claim 21, wherein a transmitting modem is at least one of a central office modem and a remote terminal modem.

28. The method of claim 21, wherein a receiving modem is at least one of a central office modem and a remote terminal modem.

29. A method for communicating data between modems using multicarrier modulation comprising:

receiving an initiate diagnostic mode message;

determining the accuracy of a received diagnostic link message; and receiving at least one of an increased transmission power diagnostic link message if the received diagnostic link message is inaccurate and a re-transmission of a predetermined number of the diagnostic link messages.

30. The method of claim 29, wherein the diagnostic link message comprises at least one of test and diagnostic information.

31. The method of claim 30, wherein the received diagnostic link message comprises at least one of a version number of a diagnostic link mode, a length of the diagnostic information, a communications standard, a chipset type, one or more vendor identifications, an ATU version number, a time domain received reverb signal, a frequency domain reverb signal, an amplifier setting, a CO transmitter power spectral density, a frequency domain received idle channel, a signal to noise ratio, bits and gain information, and upstream and downstream transmission rates.

32. The method of claim 29, wherein the accuracy is determined based on at least one of an error detecting scheme, a bit error detection and a cyclic redundancy check.

33. The method of claim 29, wherein the initiate diagnostic mode message is based on at least one of an initialization failure, a bit rate failure, a CRC error in an initialization message, a CRC error during the normal steady state transmission mode, a forward error correction error, a user request, a central office modem request and a remote terminal modem request.

34. The method of claim 29, further comprising completing a portion of a modem initialization sequence before receiving the initiate diagnostic mode message.

35. An information storage media comprising information for communicating data between modems using multicarrier modulation comprising:

information that instructs a transmitting modem to forward an initiate diagnostic mode message to a receiving modem;

information that determines a diagnostic link message;

information that transmits the diagnostic link message; and

information that determines the accuracy of the transmitted diagnostic link message.

36. An information storage media comprising information for communicating data between modems using multicarrier modulation comprising:

information that receives an initiate diagnostic mode message;

information that determines a diagnostic link message;

information that transmits the diagnostic link message; and information that at least one of increases a transmission power of the diagnostic link message if the received diagnostic link message is inaccurate and re-transmits the diagnostic link message a predetermined number of times.

37. An information storage media comprising information for communicating data between modems using multicarrier modulation comprising:

information that receives an initiate diagnostic mode message;

information that determines the accuracy of a received diagnostic link message; and

information that receives at least one of an increased transmission power diagnostic link message if the received diagnostic link message is inaccurate and a re-transmission of a predetermined number of the diagnostic link messages.

38. A method for communicating diagnostic information between DSL modems using multicarrier modulation comprising:

completing a portion of a modem initialization sequence;

transmitting an initiate diagnostic communication mode message to a receiving modem;

entering a diagnostic communications mode based on at least one of an initialization failure, a bit rate failure, a CRC error in an initialization message, a CRC error during the normal steady state transmission mode, a forward error correction error, a user request, a central office modem request and a remote terminal modem request; and

transmitting a diagnostic link message comprising at least one of a version number of a diagnostic link mode, a length of the diagnostic information, a communications standard, a chipset type, one or more vendor identifications, an ATU version number, a time domain received reverb signal, a frequency domain reverb signal, an amplifier setting, a CO transmitter power spectral density, a frequency domain received idle channel, a signal to noise ratio, bits and gain information, and upstream and downstream transmission rates.

39. The method of claim 38, further comprising re-transmitting the diagnostic link message a predetermined number of times.

40. The method of claim 38, further comprising increasing a transmission power of the diagnostic link message.

41. A method for communicating diagnostic information between DSL modems using multicarrier modulation comprising:

completing a portion of a modem initialization sequence;

receiving an initiate diagnostic communication mode message;

entering a diagnostic communications mode based on at least one of an initialization failure, a bit rate failure, a CRC error in an initialization message, a CRC error during the normal steady state transmission mode, a forward error correction error, a user request, a central office modem request and a remote terminal modem request;

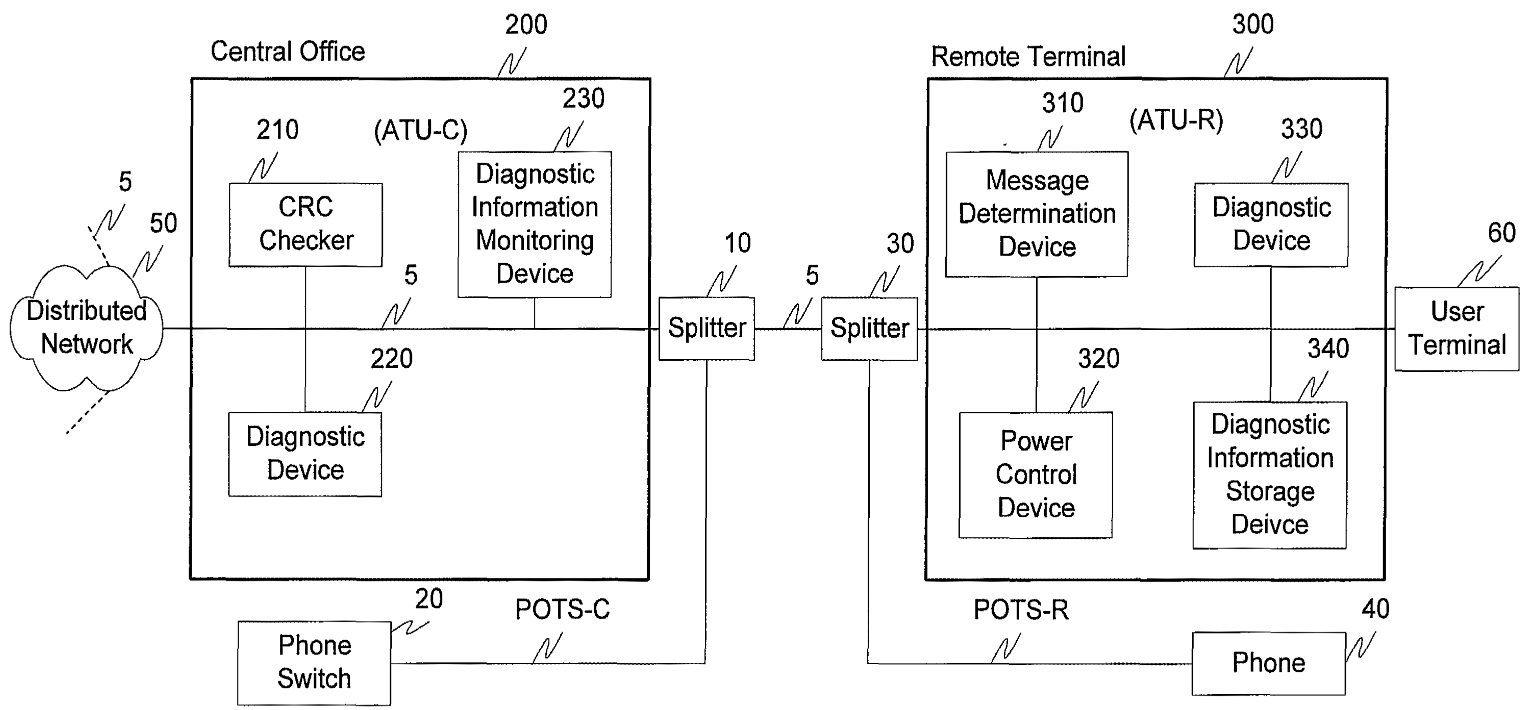
receiving a diagnostic link message comprising at least one of a version number of a diagnostic link mode, a length of the diagnostic information, a communications standard, a chipset type, one or more vendor identifications, an ATU version number, a time domain received reverb signal, a frequency domain reverb signal, an amplifier setting, a CO transmitter power spectral density, a frequency domain received idle channel, a signal to noise ratio, bits and gain information, and upstream and downstream transmission rates.

42. The method of claim 41, further comprising receiving a re-transmitted diagnostic link message a predetermined number of times.

43. The method of claim 41, further comprising receiving an increased transmission power diagnostic link message.

ABSTRACT OF THE DISCLOSURE

Upon detection of a trigger, such as the exceeding of an error threshold or the direction of a user, a diagnostic link system enters a diagnostic information transmission mode. This diagnostic information transmission mode allows for two modems to exchange diagnostic and/or test information that may not otherwise be exchangeable during normal communication. The diagnostic information transmission mode is initiated by transmitting an initiate diagnostic link mode message to a receiving modem accompanied by a cyclic redundancy check (CRC). The receiving modem determines, based on the CRC, if a robust communications channel is present. If a robust communications channel is present, the two modems can initiate exchange of the diagnostic and/or test information. Otherwise, the transmission power of the transmitting modem is increased and the initiate diagnostic link mode message re-transmitted to the receiving modem until the CRC is determined to be correct.



Central Office
200
(ATU-C)
210
CRC Checker
230
Diagnostic Information Monitoring Device
5
50
Distributed Network
5
220
Diagnostic Device
10
Splitter
5
30
Splitter
Remote Terminal
300
310
(ATU-R)
330
Message Determination Device
Diagnostic Device
60
User Terminal
320
340
Power Control Device
Diagnostic Information Storage Deivce
20
Phone Switch
POTS-C
POTS-R
40
Phone

Fig. 1

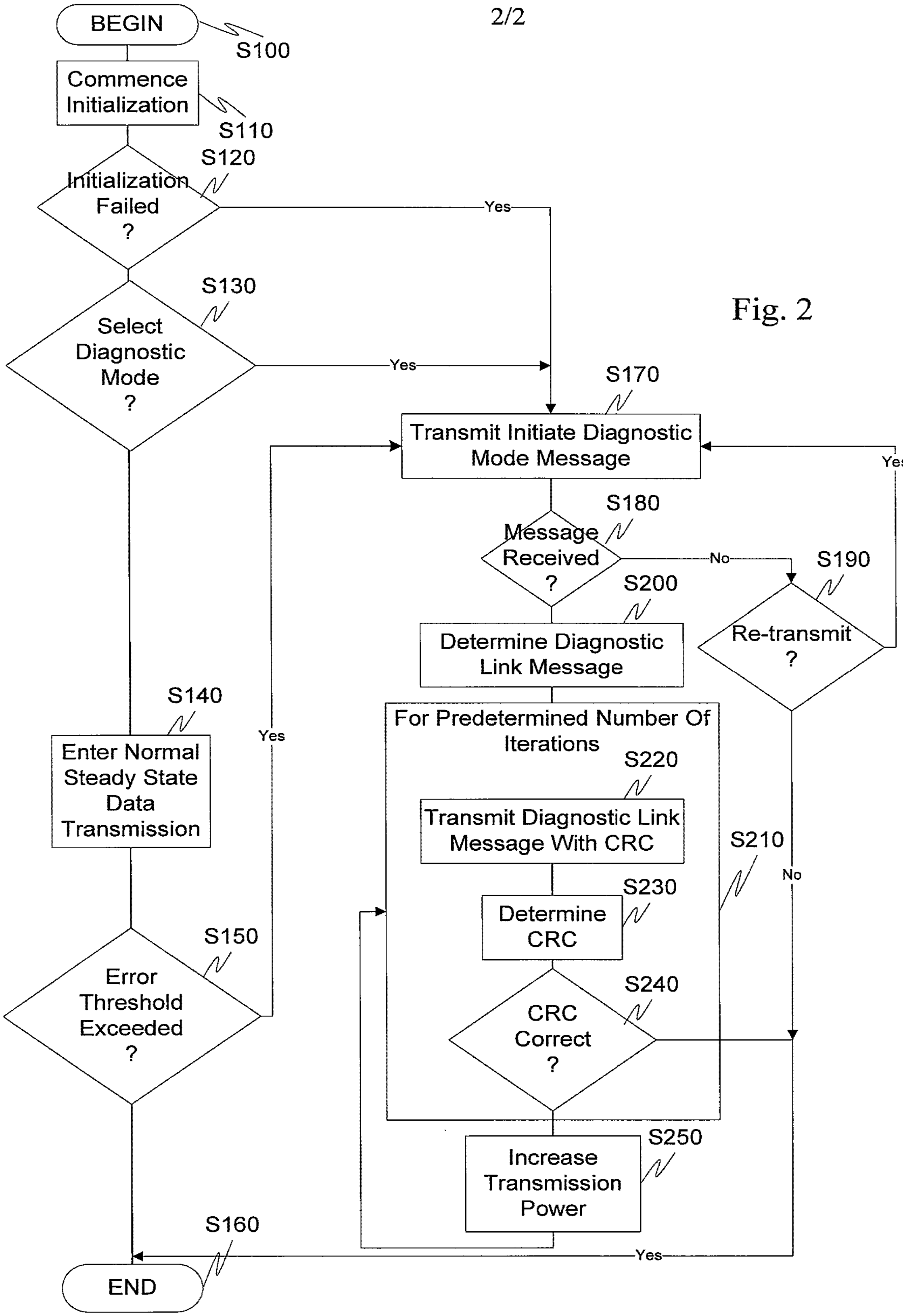

BEGIN
S100
Commence Initialization
S110
S120
Initialization Failed ?
Yes
S130
Select Diagnostic Mode ?
Yes
S170
Transmit Initiate Diagnostic Mode Message
Yes
S180
Message Received ?
No
S190
Re-transmit ?
S200
Determine Diagnostic Link Message
For Predetermined Number Of Iterations
S220
Transmit Diagnostic Link Message With CRC
S210
No
S230
Determine CRC
S240
CRC Correct ?
S250
Increase Transmission Power
S140
Enter Normal Steady State Data Transmission
Yes
S150
Error Threshold Exceeded ?
S160
Yes
END

Fig. 2

Electronic Patent Application Fee Transmittal	
Application Number:	
Filing Date:	
Title of Invention:	Multicarrier Modulation Messaging for Power Level Per Subchannel Information
First Named Inventor/Applicant Name:	David M. Krinsky
Filer:	Jason Vick/Joanne Vos
Attorney Docket Number:	5550-2-CON2-1-1
Filed as Large Entity	

Utility under 35 USC 111(a) Filing Fees

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
Utility application filing	1011	1	330	330
Utility Search Fee	1111	1	540	540
Utility Examination Fee	1311	1	220	220
Pages:				
Claims:				
Independent claims in excess of 3	1201	3	220	660
Miscellaneous-Filing:				
Petition:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**1750**

Electronic Acknowledgement Receipt

EFS ID:	7608994
Application Number:	12779660
International Application Number:	
Confirmation Number:	8981
Title of Invention:	Multicarrier Modulation Messaging for Power Level Per Subchannel Information
First Named Inventor/Applicant Name:	David M. Krinsky
Customer Number:	62574
Filer:	Jason Vick/Joanne Vos
Filer Authorized By:	Jason Vick
Attorney Docket Number:	5550-2-CON2-1-1
Receipt Date:	13-MAY-2010
Filing Date:	
Time Stamp:	17:07:09
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	Deposit Account
Payment was successfully received in RAM	$1750
RAM confirmation Number	3545
Deposit Account	191970
Authorized User	

The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows:

Charge any Additional Fees required under 37 C.F.R. Section 1.16 (National application filing, search, and examination fees)

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Charge any Additional Fees required under 37 C.F.R. Section 1.20 (Post Issuance fees)

Charge any Additional Fees required under 37 C.F.R. Section 1.21 (Miscellaneous fees and charges)

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Application Data Sheet	ADS.pdf	1104683 fc53a2c1198cc4973aa23a8dfba05f2baa0100f0	no	5

Warnings:

Information:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
2	Oath or Declaration filed	Declaration.pdf	71994 827079871c7e6133e1eb5980b8f21f9ba0a79304	no	2

Warnings:

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Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
3		AMEND_PRELIM_01.pdf	418645 61cf3753fcbda696aa69aeb57d4fff87ff822535	yes	6

Multipart Description/PDF files in .zip description

Document Description	Start	End
Preliminary Amendment	1	1
Specification	2	2
Claims	3	5
Applicant Arguments/Remarks Made in an Amendment	6	6

Warnings:

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Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
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Multipart Description/PDF files in .zip description

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Specification	1	11
Claims	12	17
Abstract	18	18

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Warnings:					
Information:					
		Total Files Size (in bytes):	1817938		

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If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

Electronic Acknowledgement Receipt

EFS ID:	7608994
Application Number:	12779660
International Application Number:	
Confirmation Number:	8981
Title of Invention:	Multicarrier Modulation Messaging for Power Level Per Subchannel Information
First Named Inventor/Applicant Name:	David M. Krinsky
Customer Number:	62574
Filer:	Jason Vick/Joanne Vos
Filer Authorized By:	Jason Vick
Attorney Docket Number:	5550-2-CON2-1-1
Receipt Date:	13-MAY-2010
Filing Date:	
Time Stamp:	17:07:09
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
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Payment was successfully received in RAM	$1750
RAM confirmation Number	3545
Deposit Account	191970
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Charge any Additional Fees required under 37 C.F.R. Section 1.21 (Miscellaneous fees and charges)

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Application Data Sheet	ADS.pdf	1104683 fc53a2c1198cc4973aa23a8dfba05f2baa0100f0	no	5

Warnings:

Information:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
2	Oath or Declaration filed	Declaration.pdf	71994 827079871c7e6133e1eb5980b8f21f9ba0a79304	no	2

Warnings:

Information:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
3		AMEND_PRELIM_01.pdf	418645 61cf3753fcbda696aa69aeb57d4fff87ff822535	yes	6

Multipart Description/PDF files in .zip description

Document Description	Start	End
Preliminary Amendment	1	1
Specification	2	2
Claims	3	5
Applicant Arguments/Remarks Made in an Amendment	6	6

Warnings:

Information:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
4		Specifciation.pdf	141780 07eee1497f235939db1420bf11e2eddc1fcae1b2	yes	18

Multipart Description/PDF files in .zip description

Document Description	Start	End
Specification	1	11
Claims	12	17
Abstract	18	18

Warnings:

Information:					
5	Drawings-only black and white line drawings	Figures.pdf	44544 47b2f618235b6b2102e2937d400af43b208125b2	no	2
Warnings:					
Information:					
6	Fee Worksheet (PTO-875)	fee-info.pdf	36292 52a4879afa8bd63987d0817eff2446d290bfa236	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	1817938		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	5550-2-CON2-1-1
	Application Number	
Title of Invention	Multicarrier Modulation Messaging for Power Level Per Subchannel Information	

The application data sheet is part of the provisional or nonprovisional application for which it is being submitted. The following form contains the bibliographic data arranged in a format specified by the United States Patent and Trademark Office as outlined in 37 CFR 1.76.
This document may be completed electronically and submitted to the Office in electronic format using the Electronic Filing System (EFS) or the document may be printed and included in a paper filed application.

Secrecy Order 37 CFR 5.2

☐ Portions or all of the application associated with this Application Data Sheet may fall under a Secrecy Order pursuant to 37 CFR 5.2 (Paper filers only. Applications that fall under Secrecy Order may not be filed electronically.)

Applicant Information:

Applicant 1 Remove

Applicant Authority ◉ Inventor | ◯ Legal Representative under 35 U.S.C. 117 | ◯ Party of Interest under 35 U.S.C. 118

Prefix	Given Name	Middle Name	Family Name	Suffix
	David	M.	Krinsky	

Residence Information (Select One) ◉ US Residency ◯ Non US Residency ◯ Active US Military Service

City	Acton	State/Province	MA	Country of Residence i	US
Citizenship under 37 CFR 1.41(b) i	US				

Mailing Address of Applicant:

Address 1	4 Ayer Road
Address 2	
City	Acton
State/Province	MA
Postal Code	01720
Country i	US

Applicant 2 Remove

Applicant Authority ◉ Inventor | ◯ Legal Representative under 35 U.S.C. 117 | ◯ Party of Interest under 35 U.S.C. 118

Prefix	Given Name	Middle Name	Family Name	Suffix
	Robert	Edmund	Pizzano	

Residence Information (Select One) ◉ US Residency ◯ Non US Residency ◯ Active US Military Service

City	Stoneham	State/Province	MA	Country of Residence i	US
Citizenship under 37 CFR 1.41(b) i	US				

Mailing Address of Applicant:

Address 1	5 Bow Street Court
Address 2	
City	Stoneham
State/Province	MA
Postal Code	02180
Country i	US

All Inventors Must Be Listed - Additional Inventor Information blocks may be generated within this form by selecting the **Add** button. Add

Correspondence Information:

Enter either Customer Number or complete the Correspondence Information section below. For further information see 37 CFR 1.33(a).

☐ **An Address is being provided for the correspondence Information of this application.**

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	5550-2-CON2-1-1
	Application Number	

Title of Invention	Multicarrier Modulation Messaging for Power Level Per Subchannel Information

Customer Number	62574	
Email Address	jvick@sheridanross.com	Add Email / Remove Email

Application Information:

Title of the Invention	Multicarrier Modulation Messaging for Power Level Per Subchannel Information		
Attorney Docket Number	5550-2-CON2-1-1	**Small Entity Status Claimed**	☐
Application Type	Nonprovisional		
Subject Matter	Utility		
Suggested Class (if any)		**Sub Class (if any)**	
Suggested Technology Center (if any)			
Total Number of Drawing Sheets (if any)	2	**Suggested Figure for Publication (if any)**	

Publication Information:

☐ Request Early Publication (Fee required at time of Request 37 CFR 1.219)

☐ **Request Not to Publish.** I hereby request that the attached application not be published under 35 U.S. C. 122(b) and certify that the invention disclosed in the attached application **has not and will not** be the subject of an application filed in another country, or under a multilateral international agreement, that requires publication at eighteen months after filing.

Representative Information:

Representative information should be provided for all practitioners having a power of attorney in the application. Providing this information in the Application Data Sheet does not constitute a power of attorney in the application (see 37 CFR 1.32).
Enter either Customer Number or complete the Representative Name section below. If both sections are completed the Customer Number will be used for the Representative Information during processing.

Please Select One:	◉ Customer Number	○ US Patent Practitioner	○ Limited Recognition (37 CFR 11.9)
Customer Number	62574		

Domestic Benefit/National Stage Information:

This section allows for the applicant to either claim benefit under 35 U.S.C. 119(e), 120, 121, or 365(c) or indicate National Stage entry from a PCT application. Providing this information in the application data sheet constitutes the specific reference required by 35 U.S.C. 119(e) or 120, and 37 CFR 1.78(a)(2) or CFR 1.78(a)(4), and need not otherwise be made part of the specification.

Prior Application Status	Pending		Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)
	Continuation of	12/477742	2009-06-03
Prior Application Status	Patented		Remove

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	5550-2-CON2-1-1
	Application Number	
Title of Invention	Multicarrier Modulation Messaging for Power Level Per Subchannel Information	

Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
12/477742	Continuation of	10/619691	2003-07-16	7570686	2009-08-04
Prior Application Status	Patented			Remove	
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
10/619691	Division of	09/755173	2001-01-08	6658052	2003-12-02
Prior Application Status	Expired			Remove	

Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)
09/755173	non provisional of	60/224308	2000-08-10
Prior Application Status	Expired		Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)
09/755173	non provisional of	60174865	2000-01-07

Additional Domestic Benefit/National Stage Data may be generated within this form by selecting the **Add** button. Add

Foreign Priority Information:

This section allows for the applicant to claim benefit of foreign priority and to identify any prior foreign application for which priority is not claimed. Providing this information in the application data sheet constitutes the claim for priority as required by 35 U.S.C. 119(b) and 37 CFR 1.55(a).

Remove

Application Number	Country i	Parent Filing Date (YYYY-MM-DD)	Priority Claimed
			○ Yes ◉ No

Additional Foreign Priority Data may be generated within this form by selecting the **Add** button. Add

Assignee Information:

Providing this information in the application data sheet does not substitute for compliance with any requirement of part 3 of Title 37 of the CFR to have an assignment recorded in the Office.

Assignee 1 Remove

If the Assignee is an Organization check here. ☒

Organization Name	AWARE, INC.		
Mailing Address Information:			
Address 1	40 Middlesex Turnpike		
Address 2			
City	Bedford	**State/Province**	MA
Country i	US	Postal Code	01730-1432
Phone Number		Fax Number	
Email Address			

Additional Assignee Data may be generated within this form by selecting the **Add** button. Add

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	5550-2-CON2-1-1
	Application Number	
Title of Invention	Multicarrier Modulation Messaging for Power Level Per Subchannel Information	

Signature:

A signature of the applicant or representative is required in accordance with 37 CFR 1.33 and 10.18. Please see 37 CFR 1.4(d) for the form of the signature.

Signature	/Jason H. Vick/			Date (YYYY-MM-DD)	2010-05-13
First Name	Jason H.	Last Name	Vick	Registration Number	45285

Privacy Act Statement

The Privacy Act of 1974 (P.L. 93-579) requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these records.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).

5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.

6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).

7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

PATENT APPLICATION

Attorney Docket No. 081513.00004

DECLARATION AND POWER OF ATTORNEY FOR PATENT APPLICATION

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name,

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled: SYSTEMS AND METHODS FOR ESTABLISHING A DIAGNOSTIC TRANSMISSION MODE AND COMMUNICATING OVER THE SAME

the specification and claims of which
☒ are attached hereto OR ☐ was filed on ________ as U.S. Application No. ________

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims.

I acknowledge the duty to disclose information which is material to the patentability as defined in Title 37, Code of Federal Regulations, §1.56.

I hereby claim priority benefits under Title 35, United States Code, §119 of any foreign or U.S. Provisional application(s) for patent listed below, and have also identified below any foreign application(s) or Provisional application(s) for patent having a filing date before that of the application on which priority is claimed:

Prior Foreign or U.S. Provisional Application(s)

(Number)	(Country)	(Day/Month/Year Filed)
60/224,308	U.S.A.	August 10, 2000
60/174,865	U.S.A.	January 7, 2000

POWER OF ATTORNEY: As a named inventor, I hereby appoint the following registered practitioners to prosecute this application and transact all business in the Patent and Trademark Office connected therewith.

Daniel W. Sixbey	20,932	Daniel S. Song	43,143
Stuart J. Friedman	24,312	Marc S. Kaufman	35,212
Charles M. Leedom, Jr.	26,477	James E. Howard	39,175
David S. Safran	27,997	Corinne R. Gorski	34,339
Thomas W. Cole	28,290	Kenneth H. Salen	43,077
Donald R. Studebaker	32,815	Jason H. Vick	45,285
Jeffrey L. Costellia	35,483	Carolyn Baumgardner	41,345
Tim L. Brackett Jr.	36,092	Luan C. Do	38,434
Eric J. Robinson	38,285		
Direct all correspondence to: [X] Customer Number 22204			

ADDRESS ALL CORRESPONDENCE TO:	DIRECT TELEPHONE CALLS TO: (name and telephone number)
Eric J. Robinson, Esq. **NIXON PEABODY LLP** **8180 Greensboro Drive** **Suite 800** **McLean, Virginia 22102**	**Eric J. Robinson** **(703) 790-9110**

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under §1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issuing thereon.

DECLARATION AND POWER OF ATTORNEY, continued

Name of sole or first inventor: David M. KRINSKY

Inventor's Signature: [signature] Date: 1/8/01

Residence: 4 Ayer Road
Acton, MA 01720

Citizenship: U.S.

Post Office Address:
(Same as above)

Name of second inventor: Robert Edmund PIZZANO, Jr.

Inventor's Signature: Robert Edmund Pizzano Jr. Date: 1/8/01

Residence: 5 Bow Street Court
Stoneham, MA 02180

Citizenship: U.S.

Post Office Address:
(Same as above)

NVA166186.1

PTO/SB/06 (07-06)
Approved for use through 1/31/2007. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

PATENT APPLICATION FEE DETERMINATION RECORD
Substitute for Form PTO-875

Application or Docket Number	Filing Date	
12/779,660	05/13/2010	☐ To be Mailed

APPLICATION AS FILED – PART I

FOR	NUMBER FILED (Column 1)	NUMBER EXTRA (Column 2)	SMALL ENTITY ☐ RATE ($)	SMALL ENTITY FEE ($)	OR	OTHER THAN SMALL ENTITY RATE ($)	OTHER THAN SMALL ENTITY FEE ($)
☒ BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A			N/A	**330**
☒ SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A			N/A	**540**
☒ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A			N/A	**220**
TOTAL CLAIMS (37 CFR 1.16(i))	12 minus 20 =	* 0	X $ =		OR	X $52 =	**0**
INDEPENDENT CLAIMS (37 CFR 1.16(h))	6 minus 3 =	* 3	X $ =			X $220 =	**660**
☐ APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $250 ($125 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).						
☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))							
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL			TOTAL	**1750**

APPLICATION AS AMENDED – PART II

AMENDMENT	05/13/2010	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	SMALL ENTITY RATE ($)	SMALL ENTITY ADDITIONAL FEE ($)	OR	OTHER THAN SMALL ENTITY RATE ($)	OTHER THAN SMALL ENTITY ADDITIONAL FEE ($)
	Total (37 CFR 1.16(i))	* 12	Minus	** 20	= 0	X $ =		OR	X $52=	0
	Independent (37 CFR 1.16(h))	* 6	Minus	***6	= 0	X $ =		OR	X $220=	0
	☐ Application Size Fee (37 CFR 1.16(s))									
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	0

AMENDMENT		CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	RATE ($)	ADDITIONAL FEE ($)		RATE ($)	ADDITIONAL FEE ($)
	Total (37 CFR 1.16(i))	*	Minus	**	=	X $ =		OR	X $ =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	X $ =		OR	X $ =	
	☐ Application Size Fee (37 CFR 1.16(s))									
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

Legal Instrument Examiner:
/STEPHEN HOOVER/