(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 1 197 064 B9

(12) **CORRECTED EUROPEAN PATENT SPECIFICATION**

Note: Bibliography reflects the latest situation

(15) Correction information:
Corrected version no 1 (W1 B1)
Corrections, see page(s) 10-14

(48) Corrigendum issued on:
02.01.2004 Bulletin 2004/01

(45) Date of publication and mention of the grant of the patent:
10.09.2003 Bulletin 2003/37

(21) Application number: **01925840.9**

(22) Date of filing: **23.04.2001**

(51) Int Cl.[7]: **H04M 11/06**, H04L 5/02, H04L 27/26

(86) International application number:
PCT/IL2001/000372

(87) International publication number:
WO 2001/082542 (01.11.2001 Gazette 2001/44)

(54) **ADSL SYSTEM FOR TRANSMISSION OF VOICE AND DATA SIGNALS**

ADSL-SYSTEM FÜR SPRACH- UND DATENÜBERTRAGUNG

SYSTEMES DE COMMUNICATION TRANSMETTANT DES SIGNAUX VOCAUX ET DE DONNEES A TRAVERS LES LIGNES TELEPHONIQUES STANDARD

(84) Designated Contracting States:
AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE TR

(30) Priority: **23.04.2000 IL 13579400**

(43) Date of publication of application:
17.04.2002 Bulletin 2002/16

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EP 1 197 064 B9

Description

FIELD OF THE INVENTION

[0001] This invention relates generally to a method and to a communication system based thereon for simultaneously conveying over standard telephone lines both voice and digital data signals, and in particular to an arrangement in which these signals are conveyed by means of an asymmetric digital subscriber line over a single twisted pair telephone cable.

BACKGROUND OF THE INVENTION

[0002] In carrying out voice communication by means of a conventional telephone network, the telephone set at the premise of each subscriber is linked by a twisted pair cable to the central office of the network. This office acts to interconnect all subscribers to the network so that they can communicate with each other via twisted pair cables.

[0003] In this modern age of information, a typical telephone subscriber has available at his residence or office not only a telephone set but also a fax machine, a desktop computer and other electronic devices capable of transmitting and receiving digital data to and from other telephone subscribers in the network as well as to Internet websites and other sites capable of processing digital data.

[0004] A multi-point to point communication system including multicarrier telephony transport is described in WO 96/24989 to Dapper *et al.* The communication system includes a hybrid fiber/coax distribution network. A head end provides for downstream transmission of telephony and control data in a first frequency bandwidth over the hybrid fiber/coax distribution network and reception of upstream telephony and control data in a second frequency bandwidth over the hybrid fiber/coax distribution network. The head end includes a head end multicarrier modem for modulating at least downstream telephony information on a plurality of orthogonal carriers in the first frequency bandwidth and demodulating at least upstream telephony information modulated on a plurality of orthogonal carriers in the second frequency bandwidth. The head end further includes a controller operatively connected to the head end multicarrier modem for controlling transmission of the downstream telephony information and downstream control data and for controlling receipt of the upstream control data and upstream telephony information. The system further includes service units, each service unit operatively connected to the hybrid fiber/coax distribution network for upstream transmission of telephony and control data in the second frequency bandwidth and for receipt of the downstream control data and telephony in the first frequency bandwidth. Each service unit includes a service unit multicarrier modem for modulating at least the upstream telephony information on at least one carrier orthogonal at the head end terminal to another carrier in the second frequency bandwidth and for demodulating at least downstream telephony information modulated on at least a band of a plurality of orthogonal carriers in the first frequency bandwidth. Each service unit also includes a controller operatively connected to the service unit multicarrier modem for controlling the modulation of and demodulation performed by the service unit multicarrier modem.

[0005] In order to be able to transform an existing telephone network which conveys voice signals over twisted pair lines into a multi-media network it is known to provide for this purpose an asymmetric digital subscriber line (ADSL). An ADSL is a point-to-point connected circuit which affords each subscriber to the network with a high-speed communication link that in addition to the usual telephone services affords many other services such as video-on-demand, conference video phone communication and high-definition TV as well as a full range of Internet services.

[0006] When an ADSL is associated with a telephone network, it then becomes necessary to place a modem at either end of the existing twisted pair telephone line conveying the voice signals. This modem serves to establish the following three channels of information:

Channel A: a high-speed downstream channel running from the central office of the network to an end user.
Channel B: a medium-speed upstream channel running from the end user to the central office.
Channel C: a conventional voice channel, commonly referred to as POTS, meaning Plain Old, Telephone Service.

[0007] In this arrangement, voice channel A is segregated from data channels A and B by bandpass filters to assure uninterrupted voice telephone service even in the event of a circuit failure in the ADSL system.

[0008] Currently available are two ADSL systems which comply with current regulatory telephone standards, namely: a split-type ADSL and a splitterless type. In an ADSL of the split-type, also known as a Full-Rate type, the voice signals which are produced concurrently with the digital data signals are split therefrom and conveyed to the central station of the network over a twisted pair cable, whereas the data signals are conveyed over another twisted pair cable.

[0009] This split is effected by a POTS splitter placed at each end of the telephone line, the splitter being provided with a low bandpass filter to permit only voice signals to be supplied to one of the twisted pair lines, and a high bandpass filter to permit only digital line signals to be fed to the other twisted pair cable.

[0010] An ADSL of the split type cannot be installed in residences or offices having the usual two-wire internal telephone line wiring. It is necessary in order to install a split-type ADSL in these facilities to provide an additional two wire line running from the modem to the splitter. Such rewiring substantially raises the expenses incurred in installing an ADSL system.

[0011] In a multiple apartment building the incoming telephone lines go into a central box in the building, from which box extend telephone lines leading to each apartment. The cost of rewiring this building to accommodate a split-type ADSL system could be considerable and possibly prohibitive.

[0012] Existing ADSL systems of the "splitterless" type which function to convey both digital data and voice signals simultaneously over a single twisted pair telephone cable include either a POTS splitter to separate the voice from the data or a POTS Line Card for this purpose. These are relatively expensive components.

[0013] In order to obviate the need to include a POTS splitter in the digital signal processor of ADSL, US Patent 5,889,856 discloses an Integrated ADSL Line Card capable of processing both data and voice signals on the same board. But while this ADSL Line Card reduces the total cost of the unit, the card is difficult to apply, for it must transmit a composite signal containing both the high-frequency digital data and low-frequency voice signals by way of the same driver circuits.

[0014] The ADSL standard, which is officially known as G992.2, describes the splitterless Asymmetric Digital Subscriber Line (ADSL) transceivers that are an interface between the telecommunications network and the customer installations and a method for transmitting data signals over a telephone subscriber wire pair. G992.2 specifies parameters and electric characteristics of ADSL transceivers.

[0015] Another limitation of existing ADSL splitterless systems is that they include an interleaving device which introduces substantial delays in the transmission of digital data.

[0016] However the greatest practical drawback of existing splitter ADSL system is that it gives the telephone subscriber only a single baseband channel for voice communication. In many cases, this single channel falls short of satisfying the subscriber's voice communication requirements.

[0017] WO 01/20864 to Tzannes describes an ADSL system and method for supporting multiple applications. A digital subscriber line system includes two transceivers in communication over a communication channel using multicarrier modulation. Application profiles are defined for characterizing transmission of information over the communication channel. Each application profile is a parameter set that is associated with a unique set of one or more applications that may become active between the transceivers and specifies the transmission requirements for such applications. Each transceiver stores the application profiles and transmits information over the communication channel according to the one of the stored application profiles. When a change in a number of applications active between the transceivers occurs, a second one of the application profiles is retrieved. The transceivers then transition to transmitting information over the communication channel according to the second application profile. The transitioning can occur without interrupting communication between the transceivers in order to retrain the transceivers.

[0018] An ADSL transceiver and method for communicating over a communications channel having a plurality of subchannels are described in WO 01/20865 to Tzannes. The transceiver is capable of dynamically switching between communicating data for a first active application set and communicating data for a second different active application set. In one embodiment the transceiver modifies the data rates of subchannels in the Bit Allocation Table according to the transmission requirements of the application whose data is being transmitted on the particular subchannel. In another embodiment, the transceiver is capable of dynamically switching from transmitting data for an Internet access application to transmitting data for a voice telephony application in addition to the Internet access application. In a further embodiment, the ABCD voice telephony signaling bits can be transmitted with either the data from the Internet access application or the data from the voice telephony application.

SUMMARY OF THE INVENTION

[0019] In view of the foregoing, the main object of this invention is to provide a method and a communication system for carrying out the method adapted to convey simultaneously voice and digital data signals over a single twisted pair telephone cable whereby an existing telephone network can without rewiring be transformed into a multi-media network.

[0020] More particularly, an object of this invention is to provide a system of the above type which includes a splitterless asymmetric digital subscriber line (ADSL) that obviates the drawbacks of prior art ADSL systems.

[0021] A significant feature of an ADSL system in accordance with the invention is that it incorporates therein a discrete multi-tone (DMT) unit producing a plurality of carriers having different frequencies, some of which are modulated by digitized voice signals, other carriers being modulated by digital data signals, which voice and data-modulated carriers are conveyed over a common twisted pair telephone line without interference therebetween.

[0022] Briefly stated, a method in accordance with the invention and a communication system based thereon for conveying simultaneously both voice and data signals via a common twisted pair line includes a splitterless ADSL having incorporated therein a discrete multitone (DMT) unit. The DMT unit yields a plurality of carriers some of which are assigned to and modulated by digitized voice signals, others being assigned to and being modulated by digital data signals.

[0023] The DMT carriers for carrying the voice signals are selected on the basis of their signal-to-noise ratio characteristics and they can be reallocated dynamically for data transmission. A first tone ordering circuit assigns data streams to data carriers, and a second tone ordering circuit assigns voice streams to voice carriers. Each carrier is modulated by QAM modulation, the data carriers by a sequence of QAM symbols representing data, the voice carriers by a sequence of QAM symbols representing voice. QAM modulation is carried out by a first constellation encoder that modulates the data carriers and a second constellation encoder that modulates the voice carriers.

[0024] Thus, in accordance with one broad aspect of the invention, there is provided a method for transmitting a voice signal and digital data over a telephone subscriber wire pair by means of a system which includes a discrete multi-tone (DMT) line signal unit producing a DMT signal constituted of a plurality of carriers of different frequency in which each carrier is modulated by a stream of QAM symbols and transmits a number of bits of information per QAM symbol correlated with the signal-to-noise ratio measured by each carrier during an initialization process, which method comprises:

A. assigning for voice transmission a portion of said plurality of carriers, said portion containing carriers each characterized by the ability to transmit a number of bits equal or larger than n per QAM symbol,

B. assigning other carriers of the DMT line signal for data transmission,

C. converting at least one voice signal into a corresponding sequence of n-bit digital words,

characterized by:

D. synchronizing said n-bit digital words with said frames of said DMT line signal,

E. encoding said sequence of n-bit digital words using a QAM constellation encoder so as to produce at least one respective sequence of n-bit QAM symbols, where each QAM symbol has a real component constituted of odd bits of said n-bit digital words and an imaginary component constituted of even bits of said n-bit digital words; and

F. loading said at least one sequence of QAM symbols in synchronization

with said sequence of n-bit digital words on respective Carriers assigned for voice transmission.

[0025] According to another broad aspect of the invention there is provided in a system for transmitting a voice signal and digital data over a telephone subscriber wire pair linked to a digital data source through a digital interface port, the system being provided with a discrete multi-tone line signal unit (DMT) producing a DMT signal constituted of a plurality of carriers having different frequencies and employing a first tone-ordering means for assigning said digital data to the carriers of the DMT signal, and employing a first constellation encoder connected to an Inverse Discrete Fourier Transformation (IDFT) processor to modulate the carriers of said DMT signal assigned to data transmission, a transmitter comprising:

at least one voice interface port connected to a corresponding external voice signal source,

a respective converter circuit connected to each voice interface port for converting said voice signal into at least one sequence of n-bit digital words,

a second tone-ordering means connected to each of the converter circuits to assign a portion of said carriers of said DMT signal for voice transmission,

a digital to analog converter coupled to said IDFT processor for converting the DMT signal to an analog signal for sending over said wire pair;

characterized by:

a synchronization circuit connected to each of the converter circuits and to said IDFT processor for synchronizing said n-bit digital words with said frames of said DMT line signal;

a second constellation encoder receiving said at least one sequence of n-bit digital words and being connected to said IDFT processor, said second constellation encoder being configured for modulating said portion of said carriers assigned to voice transmission with at least one sequence of n-bit QAM symbols that correspond to said at least one sequence of n-bit digital words, where each QAM symbol has a real component constituted of odd bits of said n-bit digital words and an imaginary component constituted of even bits of said n-bit digital words; and

a loader configured for conveying said at least one sequence of QAM symbols in synchronization with said sequence of n-bit digital words on respective carriers assigned for voice transmission.

BRIEF DESCRIPTION OF THE DRAWINGS

[0026] For a better understanding of the invention as well as other objects and features thereof, reference is made to the annexed drawings wherein:

Fig. 1 is a block diagram of a prior art ADSL system of the splitterless type;

Fig. 2 is a flow chart of the processing steps carried out in a prior art system;

Fig 3 shows the frequency spectrum of the upstream and 'downstream carriers in the prior art system;

Fig. 4 is a frequency spectrum graph showing bit loading of carriers in a system in accordance with the invention;

Fig. 5 is a time base diagram illustrating synchronization of the voice encoder with the frame rate of the ADSL system;

Fig. 6A schematically illustrates a constellation pattern of PCM words and a derived QAM vector;

Fig. 6B schematically illustrates the bit significance in a PCM word derived from a voice channel;

Fig. 7 is a block diagram showing a subcentral station and a central station in an ADSL system in accordance with a preferred embodiment of the invention;

Fig. 8A is a flow chart illustrating a sequence of data processing events in this preferred embodiment;

Fig. 8B is a flow chart illustrating a sequence of the voice processing events in this preferred embodiment;

Fig. 9 is a flow chart illustrating the sequence of the voice processing events in a multi-voice channel in the transmitter;

Fig. 10A is a flow chart illustrating the sequence of processing events in data processing in the transmitter;

Fig. 10B is a block diagram illustrating dynamic allocation of voice carriers;

Fig. 11 is the incorporation of a PCM digital telephone signal from a frame relay;

Fig. 12 schematically illustrates the incorporation of data stream into several channels of the system; and

Fig. 13 is a block diagram illustrating the incorporation of several voice channels at a subscriber's premises.

DETAILED DESCRIPTION OF THE INVENTION

Prior Art:

[0027] An ADSL system for carrying out a communication method in accordance with the invention is of the splitterless type, thereby making it possible to convey both voice and data signals over a single twisted pair telephone cable.

[0028] Inasmuch as the present system is an improvement over a prior art ADSL system of the splitterless type and uses many of the same components, the present system and its advantages over an existing system can best be understood by first considering the prior art system illustrated Figs. 1, 2 and 3.

[0029] Fig. 1 is a block diagram illustrating an existing splitterless ADSL system 101. The subscriber premises is connected to a central office 109 (CO) by a twisted wire pair telephone cable 107. At subscriber premises 103, the twisted wire pair 107 is connected to a fax machine 121, to a telephone set 123, and to a remote ATU-R 105 (ADSL Transceiver Unit), using for this purpose internal telephone lines 117. ATU-R unit 105 is connected directly to telephone line 117 and to a PC 125 (personal computer) by an Ethernet cable 124. The fax machine 121 and telephone set, 123 are connected to telephone line 117 by microfilter 119. Central Office 109 includes an ATU-C 111 (CO ADSL Transceiver Unit) a POTS-splitter 131, a POTS Line Card 108, a data switch 135, a telephone switch 137, a data network 115, and a telephone network 113.

[0030] The subscriber twisted pair 107 is coupled to POTS-splitter 131 which separates data and voice signals. The data signals pass through data switch 135 to ATU-C transceiver 111. Voice signals from voice network 113 go through a POTS Line Card 108 via telephone switch 137.

[0031] Voice signals are conveyed via telephone switch 137 and through POTS splitter 131 as baseband signals from which they are applied to twisted pair 107. Data signals conveyed through data switch 135 are modulated in a frequency range higher than that of the baseband POTS signals and are applied through POTS splitter 131 to twisted pair wire 107. Since the data communication signals are transmitted in a frequency range different from that of the voice communication signals, FDM (frequency-division-multiplexing) makes possible simultaneous transmission of both POTS signal and data communications over the same single twisted pair 107.

[0032] Fig. 2 is a flow chart of the processing steps carried out in an ATU (ADSL Transceiver Unit) referenced in ITU (International Telecommunications Union) recommendation G992.2, referring to the splitterless ADSL.

Data 153 is processed in ATM Cell Formation step 151 by an interface port resulting in a sequence of ATM (Synchronous Transfer Mode) cells. In step 155 the cells are RS (Reed - Solomon) encoded and scrambled. The ADSL system employs FEC (Forward Error Correction) based on RS encoding to reduce the effect of the impulse noise. In step 157 an interleaver mixes data bits to protect the encoded data cells from impulse noise. In step 159 tone ordering is calculated for the interleaved encoded data and the data is distributed among 128 tones (carriers) of a multitone line signal. In step 161, modulation parameters are calculated by a constellation encoder and a gain scaler for each carrier. In step 163 the modulation parameters of all carriers are transformed by an IDFT (Inverse Discrete Fourier Transformation) processing to produce digital samples of DMT (Discrete MultiTone) signals. In step 165 the digital samples are written into an output buffer. In step 167 a DTA (Digital To Analog) converter transforms the digital samples to analog DMT line signal.

[0033] The ADSL is an adaptive system. During an initialization Communication phase, an ADSL system measures the SNR (signal-to-noise ratio) for each carrier and defines the number of bits that may be loaded on the carrier.

[0034] Fig. 3 illustrates the frequency spectrum of the ADSL carriers, the figure showing a typical graph of ADSL downstream and upstream carriers (shown on the graph as bars) bit per symbol loading. In practice, for cable length of up to 9000 feet, many carriers may be loaded with a high number of bits per symbol (up to 12

- 14 bits).

[0035] As previously pointed out, an existing ADSL system of the type shown in Fig. 1 has several drawbacks. The most serious of which is that the system provides the telephone subscriber with only a single baseband voice channel which for many subscribers is inadequate. And the inclusion of a POTS Line Card in this prior art system adds substantially to its cost.

The Invention: In a method in accordance with the invention and in a splitterless ADSL system for carrying out this method to convey voice and data signals simultaneously over single twisted pair telephone cable, the disadvantage of prior art systems are overcome, particularly in regard to voice transmission. Instead of a single voice channel, an ADSL system in accordance with the invention has inserted therein several high quality telephone channels.

[0036] The multi-tone modulation technique included in a system in accordance with the invention acts to separate the available bandwidth into a multiplicity of distinct carriers, each functioning as a communication channel. This makes it possible to convey voice and digital data signals simultaneously on different channels. In practice the DMT-ADSL splitterless system for short distances such as 9000 feet can support up to 8 telephone channels upstream with a bit rate of up to 250 Kb/s, and downstream with a bit rate of up to 6 Mb/s.

[0037] Fig. 4 illustrates the frequency spectrum of a typical graph of ADSL downstream and upstream carrier bit-loading in a system in accordance with the invention. In the upper graph in Fig. 4, the frequency of the upstream carriers is plotted against the maximum bits per symbol. In the lower graph, the frequency of the downstream carriers is plotted against the maximum bits per symbol.

[0038] Two upstream carriers and two downstream carriers are utilized for one voice channel and are hereinafter called "voice carriers" (VCs). The VCs are not determined before the onset of communication. During an initialization process, the ADSL system measures the (SNR) signal-to-noise ratio for each carrier and defines the number of bits that may be loaded on respective carriers. Two downstream and two upstream carriers having the highest SNR which are able to carry more than 8-bits are then assigned for voice transmission. The selected carriers can carry more than 8 bits for each symbol, nevertheless they are only loaded with 8 bit symbols, as can be seen in black shading in the graph.

[0039] The assignment of VCs does not interfere with the ADSL function because the working standard does not imply a definite number of carriers for use in data transmission, a situation which likewise allows disabling of some carriers with low SNR by the ADSL transmitter during the initialization process. In accordance with the present invention VCs assigned by the ADSL system are not used for data transmission. The ADSL system can therefore utilize the other DMT carriers for data transmission in accordance with G.992.2 standard.

[0040] Before being transmitted, a voice signal is converted by a PCM (Pulse Code Modulation) encoder into 8-bit digital words with frequency of 8 kHz in accordance with existing standard for PCM telephone system (ANSI T1 or E1)

[0041] Referring to Fig. 5 shown therein is a time base diagram illustrating the correlation between the PCM encoded voice signal and the DMT frames. Because the sampling rate of the PCM encoder is 8 kHz and the DMT-ADSL frames shown in row A operate at a frequency of 4 kHz, it is necessary to use two upstream VCs and two downstream VCs for transmitting one telephone channel. In row B, voice signal samples 2 and 3 are assigned to two different VCs in row C1 and row C2 which correspond exactly to the DMT-ADSL flames in row A. The PCM encoder is therefore synchronized with the DMT-ADSL frame rate.

[0042] Each carrier of the DMT-ADSL system undergoes modulation by a sequence of QAM symbols. This is achieved by operating a QAM constellation encoder. Each 8-bit PCM word is transformed into an 8-bit QAM vector, following which odd QAM symbols modulate a first VC, and even QAM symbols modulate a second VC.

[0043] Fig. 6A illustrates the derivation of a QAM vector in the context of a constellation encoder according to the present invention. The constellation encoder calculates real and imaginary components of the QAM vector using odd bits of the PCM word for the real component of the vector, and even bits of the PCM word for the imaginary component of this vector.

[0044] As can be seen in Fig. 6B, more significant bits of a PCM word correspond to the more significant bits of real and imaginary components of the respective QAM vector. Errors may be produced by channel noise only in the less significant bits of the PCM words because of the short distance between consecutive QAM vectors. As a result, errors in low significant bits of PCM words produce only small additional noise in the voice signal. As a consequence, the present invention promotes a high quality voice signal transmission over an ADSL system without implementing error correction coding and without significant delay. An ADSL communication system in accordance with the invention may be extended to effect simultaneous transmission of several voice channels.

[0045] <u>**The Basic System**</u>**: Fig. 7** illustrates schematically a communication system 201 in accordance with a preferred embodiment of the present invention. A subscriber premises 103 is coupled to the CO (central office) 109 of a telephone network by a twisted wire pair telephone cable 107. At the subscriber's premises 103, the twisted wire pair 107 is connected to ATU-R transceiver unit 205. A fax machine 121 and a telephone set 123 are connected to a voice interface port 203 of ATU-R 205, using for this purpose internal telephone lines 117. A PC 125 (personal computer) is connected to a digital interface port 204 of ATU-R 205 by an Ethernet cable 124.

[0046] Central Station 109 contains an ATU-C transceiver unit 211, a data switch 135, a telephone switch 137, a data network 115, and a telephone network 113. A subscriber twisted pair cable 107 is coupled directly to ATU-C 211 and carries data and voice on (DMT) discrete multi-tone line carriers. Data signals flow from data interface port 209 of ATU-C 211 to the data switch 135. Voice signals flow from voice interface port 207 of ATU-C 211 to telephone switch 137. Telephone switch 137 is coupled to telephone network 113 whereas data switch 135 is coupled to data network 115.

[0047] The communication system 201 shown in Fig. 7 does not entail an expensive POTS-splitter nor a POTS Line Card. In accordance with the present invention, voice signals are transmitted in digital form using a portion of the capacity of the ADSL link. System 201 uses for transmitting voice and data signals different carriers (VCs and "data carriers" respectively) of the DMT-ADSL line signal.

[0048] Data Processing: **Fig. 8A** is a flow chart of the data processing events implemented in an ATU transmitter (ADSL Transceiver Unit) according to the present invention. Data 153 is processed in step 151 by an interface port resulting in a sequence of ATM (Asynchronous Transfer Mode) cells. In step 155 these cells are scrambled and RS encoded. In step 157, an interleaver mixes data bits to protect the encoded blocks of data from impulse noise. In step 263 tone ordering is calculated for the interleaved encoded data and the data is distributed to 126 tones or carriers of the multitone line signaL

[0049] In step 265, modulation parameters are calculated by a constellation encoder and a gain scaler for each data carrier. In step 267 the modulation parameters of all carriers are transformed by IDFT (Inverse Discrete Fourier Transformation) processing to produce digital samples of the DMT (Discrete MultiTone) signal. In step 269 the digital samples are wetter into an output buffer. In step 272 a DTA (Digital To Analog) converter transforms the digital samples to an analog DMT line signal.

[0050] Voice signal 251 undergoes a different set of processing steps. Fig. 8B shows schematically the sequence of these processing events within the ATU. In step 252, the signal is amplified and filtered by a voice interface port. In step 257 the voice signal is transformed by PCM encoding into a 64 kbit/sec sequence of 8-bit PCM words, the sampling rate of the PCM encoder is 8 KHz. It uses for this purpose standard A-Law or μ-Law coding, identical to the routine used by the PCM telephone systems ANSI T1 or E1. In step 259, tone ordering is calculated for the PCM word stream, and the PCM stream is distributed between two VCs of the DMT signal. In step 261 each 8-bit PCM word is transformed into one 8-bit QAM symbol by a VCs constellation encoder and gain-scaler. A fixed 8-bit loading on each "voice carrier" is then provided. In step 355 the sampling of the PCM encoder is synchronized with the frames of the DMT line signal.

[0051] **Data and multiple voice channels:** An ADSL transmitter in accordance with the invention can support a number of telephone channels. Data is processed in the ATU transmitter along the same lines set forth in Fig. 8A. Data 153 is processed in step 151 by an interface port resulting in a sequence of ATM cells. In step 155 the cells are scrambled and RS encoded. In step 157 an interleaver mixes data bits to protect the encoded blocks of data from impulse noise. In step 263 tone ordering is calculated for the interleaved encoded data and the data is distributed to data carriers of the multitone line signal. In step 265 modulation parameters are calculated by a constellation encoder and a gain scaler for each carrier. In step 267 the modulation parameters of all the carriers are transformed by IDFT processing to produce digital samples of DMT (Discrete MultiTone) signal. In step 269 the digital samples are written into an output buffer. In step 272 a DTA converter transforms the digital samples to an analog DMT line signal.

[0052] Fig. 9 illustrates the sequence of processing events which a voice signal that is one of several incoming signals undergoes in the ATU transmitter, starting with a voice signal 251A flowing into an individual interface port In step 252, the signal is amplified and filtered by one of the interface ports available. In step 257 the voice signal is transformed by PCM encoding into a 64 kbit/sec sequence of 8-bit PCM words, the sampling rate of the PCM coder is 8kHz using for this purpose standard A-Law or μ-Law coding, much the same as is used by the PCM telephone systems ANSI T1 or E1.

[0053] In step 259, tone ordering is calculated for the PCM word stream, and each 64 kb/sec PCM stream is distributed between two VCs of DMT signal. In step 260 parity bytes are calculated by an RS encoder for PCM words of all the voice channels, which are then loaded on an additional "voice carrier". In step 261 each 8-bit PCM word is transformed into one 8-bit QAM symbol by a VCs constellation encoder and gain-scaler. A fixed 8-bit loading on each "voice carrier" is consequently provided. In step 355 the sampling of the PCM encoder is synchronized with the frames of the DMT line signal.

[0054] **Data communication over silent voice carriers: Fig. 10A** illustrates the sequence of processing events involved in data processing within the ATU transmitter in accordance with another embodiment of the present invention. Data 153 is processed in step 151 by an interface port resulting in a sequence of ATM cells. In step 155 the cells are scrambled and RS encoded. In step 157 an interleaver mixes data bits to protect the encoded blocks of data from impulse noise. In step 380, the interleaved data stream is distributed between "data carriers" and silent VCs.

[0055] To illustrate the way in which the allocation of data carriers is interactively changed, reference is now made to the embodiment of the invention shown in Fig. 10B which illustrates the voice processing block layout in a dynamic VCs allocating procedure. A carrier allocator 301 receives allocation instructions from a processor 303. This processor conducts periodic analyses of the voice signals at voice interface port 253A. Upon the identification of a silent voice channel, it instructs carrier allocator 301 to reassign the respective VCs to data communication, in addition to those assigned to data exclusively. Interface ports 253A and the others, amplify and filter the corresponding voice signals respectively. PCM encoders such as 257A sample the voice signal coming from the corresponding voice interface port at a sampling rate of the 8 kHz, transforming the analog voice signal into a 64-kbit/sec sequence of 8-bit PCM words. Each PCM encoder uses standard A-Law or μ-Law coding, the same used in PCM telephone systems ANSI T1 or E1. The PCM words of active voice channels pass then to carrier allocator 301 and then to VCs tone-ordering block 259 that distributes each 64 kbit/sec PCM word stream of busy telephone channels between two VCs of DMT signal. Carrier allocator 301 passes on to the "voice carriers" tone ordering block 259 only those PCM signals originating from such PCM encoders that are connected to active voice interface ports. Inactive PCM coders that are not currently in use by telephone lines are not connected to "voice carriers" tone ordering block 259.

[0056] An RS encoder calculates parity bytes for PCM words of active voice channels and puts these parity bytes on an additional "voice carrier". A VCs constellation encoder and gain-scaler transforms each 8-bit PCM word into one 8-bit QAM symbol and provides fixed 8-bit loading on each "voice carrier". The ampling rate of each PCM coder is synchronized with the frame rate of DMT line signal.

[0057] A communication system in accordance with this embodiment of the invention can potentially transmit more data because a portion of the telephone channels which is not busy, for example during off-peak hours, can be utilized for communicating data coming from data interleaver 357.

[0058] **<u>Incorporating a digital voice channel of the CO</u>:** An ATU-C transmitter in accordance with the invention is well adapted to incorporate electronic communication equipment of the CO, such as a PCM telephone switch (flame relay). According to a preferred embodiment of the invention, a stream of PCM telephone words of the CO is readily processed and communicated through the ADSL system. Data is processed in the same way as in example 1. Fig. 11 to which reference is now made, illustrates schematically the incorporation of a PCM digital telephone signal from a frame relay. A telephone signal comes in from frame relay 282 in the form of a 64 kbit/s PCM stream. It is sent to an input of a PCM interface 283 that synchronizes the DMT signal frames with the 8-bit PCM words. The 64kb/s PCM stream is then distributed between two VCs of DMT signal as in a previous example. To synchronize the PCM stream with the DMT signal frames, a main 8-kHz clock 285 of the frame relay is connected to the synchronizer 255 and to the PCM interface 283. The synchronizer 255 is also connected to the ADSL 400 to synchronize between the data and the T1 voice source.

[0059] **<u>Incorporating several digital voice channels of the CO</u>:** An ATU-C transmitter in accordance with the invention is well adapted to incorporate electronic communication equipment of the CO, such as PCM telephone switch (frame relay) having an ANSI T1 interface. According to a preferred embodiment of the invention, several streams of PCM telephone words of the CO are readily processed and communicated through the ADSL system.

[0060] Data is processed in the same way as in the first example. Fig. 12 illustrates schematically the incorporation of a T1 format data stream containing several digital telephone channels into the ADSL system. First, the data stream 271 coming from a frame relay in T1 format is split into several channels by a T1 interface 277. Each such channel carries a sequence of 8 bit PCM words at a bit rate of 64 kb/s of a respective telephone channel. In the next step, each PCM stream 279 A, B,... is modulated by VCs DMT modulator 259 that distributes each 64 kb/sec PCM stream between two VCs of DMT signal. In the next step, a VCs QAM modulator and gain-scaler 261 transforms each 8-bit PCM word into one 8-bit QAM symbol and provides a fixed 8-bit loading on each one of the VCs. A synchronization block 255 synchronizes T1 system clock with the frames of the DMT line signal.

[0061] **<u>Incorporation of several voice channels at a subscriber's premises</u>:** Data is processed and transmitted in an ATU-R in the same way as described in the first example. Referring now to Fig. 13, it will be seen that voice channel 251A is connected to a voice interface port 253A which is one of several identical ports, where the necessary amplifying and filtering is performed. A PCM encoder 257A is connected to the respective voice interface ports 253A. Each PCM of the encoders has a sampling rate of 8 kHz and transforms an analog voice signal into a 64-kbit/sec sequence of 8-bit PCM words. The PCM coders use standard A-Law or μ-Law coding, which is the same one used in PCM telephone systems T1 or E1.

[0062] All PCM encoders are connected to a PCM concentrator 309. The output of concentrator 309 is connected to a VCs tone-ordering block 259 that distributes each 64 kbit/sec PCM stream between two VCs of DMT signal. A single PCM concentrator 309 is able to support several telephone channels simultaneously. Thus, for example, 8 PCM coders may be connected to one PCM concentrator that uses only four voice carriers to provide two telephone channels simultaneously.

[0063] While there has been described and illustrated methods for simultaneously conveying both data and voice sequences over a twisted pair telephone line and various systems for carrying out these methods, it must be understood that many changes may be made thereon without departing from the scope of the invention as defined by the appended claims.

Claims

1. A method for transmitting a voice signal and digital data over a telephone subscriber wire pair by means of a system which includes a discrete multitone, hereinafter referred to as DMT, line signal unit producing a DMT signal constituted of a plurality of carriers of different frequency in which each carrier is modulated by a stream of QAM symbols and transmits a number of bits of QAM symbol information correlated with the signal-to-noise ratio measured by each carrier during an initialization process, which method comprises:

    A. assigning for voice transmission a portion of said plurality of carriers, said portion containing carriers each **characterized by** the ability to transmit a number of bits equal or larger than n per QAM symbol,
    B. assigning other carriers of the DMT line signal for data transmission,
    C. converting at least one voice signal into a corresponding sequence of n-bit digital words, **characterized by:**
    D. synchronizing said n-bit digital words with said frames of said DMT line signal,
    E. encoding said sequence of n-bit digital words using a QAM constellation encoder so as to produce at least one respective sequence of n-bit QAM symbols, where each QAM symbol has a real component constituted of odd bits of said n-bit digital words and an imaginary component constituted of even bits of said n-bit digital words; and
    F. bit-loading said at least one sequence of QAM symbols in synchronization with said sequence of n-bit digital words on respective carriers assigned for voice transmission.

2. A method according to Claim 1 wherein several carriers of said assigned portion of carriers may be reassigned for data carrying when upon analysis the respective voice channels are identified as silent.

3. A method according to Claim 1 and wherein said n-bit digital words and said n-bit QAM symbols are 8-bit integers respectively.

4. A method according to Claim 1 and wherein said at least one voice signal is converted to a respective sequence of n-bit digital words by PCM encoding.

5. A method according to Claim 1 wherein more significant bits of the n-bit digital words correspond to the more significant bits of the real and imaginary components of the QAM symbols.

6. A method according to Claim 1 and wherein said at least one voice signal is associated with at least one respective telephone channel in analog or in digital form.

7. A method according to Claim 1 and wherein several carriers of said assigned portion of carriers may be reassigned for data carrying when upon analysis of a telephone signaling the respective telephone channels are identified as silent.

8. A transmitter for use in a system for transmitting a voice signal and digital data over a telephone subscriber wire pair linked to a digital data source through a digital interface port, the system being provided with a discrete multi-tone, hereinafter referred to as DMT, line signal unit, producing a DMT signal constituted of a plurality of carriers having different frequencies and employing a first tone-ordering means for assigning said digital data to the carriers of the DMT signal, and employing a first constellation encoder connected to an Inverse Discrete Fourier Transformation, hereinafter referred to as IDFT, processor to modulate the carriers of said DMT signal assigned to data transmission, the transmitter comprising:

    at least one voice interface port connected to a corresponding external voice signal source,
    a respective converter circuit connected to each voice interface port for converting said voice signal into at least one sequence of n-bit digital words,
    a second tone-ordering means connected to each of the converter circuits to assign a portion of said carriers of said DMT signal for voice transmission,
    a digital to analog converter coupled to said IDFT processor for converting the DMT signal to an analog signal for sending over said wire pair;

characterized by:

    a synchronization circuit connected to each of the converter circuits and to said IDFT processor for synchronizing said n-bit digital words with said frames of said DMT line signal;
    a second constellation encoder receiving said at least one sequence of n-bit digital words and being connected to said IDFT processor, said second constellation encoder being configured for modulating said portion of said carriers assigned to voice transmission with at least one sequence of n-bit QAM symbols that correspond to said at least one sequence of n-bit dig-

ital words, where each QAM symbol has a real component constituted of odd bits of said n-bit digital words and an imaginary component constituted of even bits of said n-bit digital words; and

a a bit loader configured for conveying said at least one sequence of QAM symbols in synchronization with said sequence of n-bit digital words on respective carriers assigned for voice transmission.

9. A transmitter according to Claim 8 wherein more significant bits of the n-bit digital words correspond to the more significant bits of the real and imaginary components of the QAM symbols.

10. A transmitter according to Claim 8 wherein said at least one external voice source is at least one analog telephone channel and said least one converter circuit is at least one pulse code modulation, hereinafter reference to as PCM, encoder, and wherein at least one voice interface port is connected to said at least one PCM encoder respectively to transform said at least one voice signal of said at least one telephone channel into at least one corresponding sequence of n-bit PCM words, and wherein said at least one PCM encoder is connected to said second tone ordering means, and wherein said at least one PCM encoder and said IDFT processor are connected to a synchronizer for synchronizing said DMT frames with said at least one sequence of n-bit digital words of said PCM encoder.

11. A transmitter according to Claim 8 and wherein said at least one voice signal arrives through at least one external telephone channel connected to said at least one voice interface port.

12. A transmitter according to Claim 8 and wherein a processor for analyzing voice at said at least one voice interface port, is connected to a carrier allocator, sending it instructions to reassign carriers formerly assigned to voice transmission, to data transmission upon the identification of corresponding voice channel as being silent.

13. A transmitter according to Claim 8 and wherein said subscriber wire pair is a twisted wire pair.

14. A transmitter according to Claim 10 and wherein a plurality of voice interface ports are connected to a corresponding number of PCM encoders, and wherein a PCM concentrator is connected to said PCM encoders and to said second tone ordering means.

15. A transmitter according to Claim 11 and wherein a processor for analyzing telephone control signals at said telephone channel, is connected to a carrier allocator, sending instructions to reassign carriers formerly assigned to voice transmission to data transmission upon the identification of corresponding telephone channel as being silent.

16. A transmitter according to Claim 11 and wherein said at least one voice interface port is a PCM voice interface port connected to an external PCM telephone channel equipment of a telephone station, to said second tone ordering means and to a synchronizer connected to said IDFT processor for synchronizing said DMT frames with said at least one sequence of n-bit digital words of said external PCM telephone channel equipment.

Patentansprüche

1. Verfahren zum Übertragen eines Sprachsignals sowie digitaler Daten über ein Teilnehmerleitungspaar mit Hilfe eines Systems mit einer diskreten Multiton-Leitungssignaleinheit, nachstehend bezeichnet als DMT, die ein aus mehreren Trägern unterschiedlicher Frequenz bestehendes DMT-Signal erzeugt, in dem jeder Träger durch einen Strom von QAM-Symbolen moduliert wird und eine Anzahl von QAM-Symbol-Informationsbits überträgt, die mit dem von jedem Träger während eines Initialisierungsprozesses gemessenen Signal-Rausch-Abstand korreliert sind, mit den folgenden Schritten:

A. Zuweisen eines Teils der Anzahl von Trägern für die Sprachübertragung, wobei dieser Teil Träger enthält, die jeweils durch die Fähigkeit zur Übertragung einer Bitanzahl **gekennzeichnet** sind, welche gleich n pro QAM-Symbol oder größer ist;

B. Zuweisen weiterer Träger des DMT-Leitungssignals zur Datenübertragung;

C. Umwandlung wenigstens eines Sprachsignals in eine entsprechende Folge digitaler n-Bit-Wörter; **gekennzeichnet durch:**

D. Synchronisieren der digitalen n-Bit-Wörter mit den Rahmen des DMT-Leitungssignals;

E. Kodieren der Folge digitaler n-Bit-Wörter unter Einsatz eines QAM-Konstellations-Codierers in der Weise, dass wenigstens eine entsprechende Folge von n-Bit-QAM-Symbolen erzeugt wird, wobei jedes QAM-Symbol einen aus ungeradzahligen Bits der digitalen n-Bit-Wörter bestehenden Realteil und einen aus geradzahligen Bits der digitalen n-Bit-Wörter bestehenden Imaginärteil aufweist; und

F. Bit-Laden der wenigstens einen Folge von

QAM-Symbolen in Synchronisation mit der Folge digitaler n-Bit-Wörter auf zur Sprachübertragung zugewiesenen entsprechenden Trägern.

2. Verfahren nach Anspruch 1, wobei mehrere Träger des zugewiesenen Trägeranteils für das Führen von Daten neu zugewiesen werden können, wenn bei der Analyse die entsprechenden Sprachkanäle als stumm erkannt werden.

3. Verfahren nach Anspruch 1, wobei die digitalen n-Bit-Wörter bzw. QAM-Symbole 8-Bit-Ganzzahlen sind.

4. Verfahren nach Anspruch 1, wobei das wenigstens eine Sprachsignal durch PCM-Codierung in eine entsprechende Folge digitaler n-Bit-Wörter umgewandelt wird.

5. Verfahren nach Anspruch 1, wobei signifikantere Bits der digitalen n-Bit-Wörter den signifikanteren Bits der Real- und Imaginärteile der QAM-Symbole entsprechen.

6. Verfahren nach Anspruch 1, wobei wenigstens ein Sprachsignal wenigstens einem entsprechenden Fernsprechkanal in analoger oder digitaler Form zugeordnet ist.

7. Verfahren nach Anspruch 1, wobei mehrere Träger des zugewiesenen Trägeranteils zur Datenführung neu zugewiesen werden können, wenn bei der Analyse eines mit Signalen beaufschlagenden Telefons die entsprechenden Fernsprechkanäle als stumm erkannt werden.

8. Sender zum Einsatz in einem System zur Übertragung eines Sprachsignals und digitaler Daten über ein mit einer digitalen Datenquelle über einen digitalen Schnittstellenport verbundenes Teilnehmerleitungspaar, wobei das System mit einer diskreten Multiton-Leitungssignaleinheit, nachfolgend bezeichnet als DMT, versehen ist, die ein aus mehreren Trägern unterschiedlicher Frequenz bestehendes DMT-Signal erzeugt sowie eine erste Tonordnungseinrichtung zum Zuweisen der digitalen Daten an die Träger des des DMT-Signals und einen mit einem inversen diskreten Fourier-Transformations-Prozessor (nachstehend bezeichnet als IDFT) verbundenen ersten Konstellationscodierer zum Modulieren der Träger des für die Datenübertragung zugewiesenen DMT-Signals benutzt, wobei der Sender aufweist:

wenigstens einen mit einer entsprechenden externen Sprachsignalquelle verbundenen Sprachen-Schnittstellenport;
eine mit jeden Sprach-Schnittstellenport verbundene entsprechende Umsetzerschaltung zur Umwandlung des Sprachsignals in wenigstens eine Folge digitaler n-Bit-Wörter;
eine mit jeder der Umsetzerschaltungen verbundene zweite Tonordereinrichtung für die Zuweisung eines Trägerteils des DMT-Signals zur Sprachübertragung;
einen mit dem IDFT-Prozessor verbundenen D/A-Wandler zum Umsetzen des DMT-Signals in ein über das Leitungspaar auszusendendes Analogsignal;

gekennzeichnet durch:

eine mit jeder der Konverterschaltungen und mit dem IDFT-Prozessor verbundene Synchronisierschaltung zum Synchronisieren der digitalen n-Bit-Wörter mit den Rahmen des DMT-Leitungssignals;
einen wenigstens eine Folge digitaler n-Bit-Wörter empfangenden und mit dem IDFT-Prozessor verbundenen zweiten Konstellationscodierer, der zum Modulieren des der Sprachübertragung zugeordneten Trägerteils mit wenigstens einer Folge von der wenigstens einen Folge digitaler n-Bit-Wörter entsprechenden n-Bit-QAM-Symbolen konfiguriert ist, wobei jedes QAM-Symbol einen aus ungeradzahligen Bits der digitalen N-Bit-Wörter bestehenden Realteil und einen aus geradzahligen Bits der digitalen n-Bit-Wörter bestehenden Imaginärteil aufweist; und
einen zum Befördern der wenigstens einen Folge von QAM-Symbolen in Synchronisation mit der Folge digitaler n-Bit-Wörter auf der Sprachübertragung zugeordneten entsprechenden Trägern konfigurierten Bitlader.

9. Sender nach Anspruch 8, wobei signifikantere Bits der digitalen n-Bit-Wörter den signifikanteren Bits der Real- und Imaginärteile der QAM-Symbole entsprechen.

10. Sender nach Anspruch 8, wobei die wenigstens eine externe Sprachquelle wenigstens ein analoger Fernsprechkanal und die wenigstens eine Umsetzerschaltung wenigstens ein Pulscodemodulations-Codierer, nachfolgend als PCM bezeichnet, ist, und wobei wenigstens ein Sprachschnittstellenport mit dem wenigstens einen PCM-Codierer verbunden ist, um das wenigstens eine Sprachsignal des wenigstens einen Fernsprechkanals in wenigstens eine entsprechende Folge von n-Bit-PCM-Wörtern umzuwandeln, wobei der wenigstens eine PCM-Codierer mit der zweiten Tonordnungseinrichtung verbunden ist, und wobei der wenigstens eine PCM-Codierer und der IDFT-Prozessor zum Synchronisieren der DMT-Rahmen mit der wenigstens einen Folge digitaler n-Bit-Wörter des PCM-Codierers mit einer Synchronisiereinrichtung verbunden sind.

11. Sender nach Anspruch 8, wobei das wenigstens eine Sprachsignal über wenigstens einen mit dem wenigstens einen Sprachschnittstellenport verbundenen Fernsprechkanal ankommt.

12. Sender nach Anspruch 8, wobei ein Prozessor zur Sprachanalyse an dem wenigstens einen Sprachschnittstellenport mit einem Trägerzuordner verbunden ist und Befehle zur Neuzuweisung von früher der Sprachübertragung zugeordneten Trägern für die Datenübertragung an diesen gibt, wenn der entsprechende Sprachkanal als stumm erkannt wird.

13. Sender nach Anspruch 8, wobei das Teilnehmerleitungspaar ein verdrilltes Leitungspaar ist.

14. Sender nach Anspruch 10, wobei eine Anzahl von Sprachschnittstellenports mit einer entsprechenden Anzahl von PCM-Codierern und ein PCM-Konzentrator mit den PCM-Codierern und der zweiten Tonordnungseinrichtung verbunden sind.

15. Sender nach Anspruch 11, wobei ein Prozessor zum Analysieren von Fernsprechsteuersignalen am Fernsprechkanal mit einem Trägerzuordner verbunden ist und Befehle zur Neuzuweisung von früher der Sprachübertragung zugeordneten Trägern für die Datenübertragung schickt, wenn der entsprechende Fernsprechkanal als stumm erkannt wird.

16. Sender nach Anspruch 11, wobei der wenigstens eine Sprachschnittstellenport ein PCM-Sprachschnittstellenport ist, der mit einer externen PCM-Fernsprecheinrichtung einer Fernsprechstelle, der zweiten Tonordnungseinrichtung und einer mit dem IDFT-Prozessor verbundenen Synchronisiereinrichtung zum Synchronisieren der DMT-Rahmen mit der wenigstens einen Folge digitaler n-Bit-Wörter der externen PCM-Fernsprechkanaleinrichtung verbunden ist.

Revendications

1. Un procédé pour émettre un signal vocal et des données numériques vers une paire de câbles d'un abonné au téléphone aux moyens d'un système qui comprend une unité de signal de ligne à multitonalité discrète , désignée ci-après sous le nom de DMT, produisant un signal constitué d'une pluralité de porteuses de fréquences différentes dans lequel chaque porteuse est modulée par un train de symboles QAM et émet un certain nombre de bits d'information de symboles QAM corrélés avec le rapport signal à bruit mesuré par chaque porteuse au cours d'un processus d'initialisation, lequel procédé comprend :

A. une attribution, pour la transmission vocale, d'une partie de ladite pluralité de porteuses, ladite partie contenant des porteuses dont chacune est **caractérisée par** la possibilité d'émettre un nombre de bits égal ou supérieur à n par symbole QAM,
B. une attribution d'autres porteuses du signal de ligne DMT pour la transmission des données,
C. une conversion d'au moins un signal vocal en une séquence correspondante de mots numériques de n bits,
caractérisé par :
D. une synchronisation desdits mots numériques de n bits avec lesdites trames dudit signal de ligne DMT,
E. un codage de ladite séquence de mots numériques de n bits en utilisant un codeur de constellation QAM afin de produire au moins une séquence respective de symboles QAM de n bits, où chaque symbole QAM possède une composante réelle constituée des bits impairs desdits mots numériques de n bits et une composante imaginaire constituée des bits pairs desdits mots numériques de n bits; et
F. un chargement de bits de ladite au moins une séquence de symboles QAM en synchronisation avec ladite séquence de mots numériques de n bits sur les porteuses respectives attribuées à la transmission vocale.

2. Un procédé selon la revendication 1, dans lequel plusieurs porteuses de ladite partie attribuée de porteuses, peuvent être attribuées de nouveau pour acheminer des données quand, suite à une analyse, les voies vocales respectives sont identifiées comme étant silencieuses.

3. Un procédé selon la revendication 1, dans lequel lesdits mots numériques de n bits et lesdits symboles QAM de n bits sont des nombres entiers de 8 bits respectivement.

4. Un procédé selon la revendication 1, dans lequel ledit au moins un signal vocal est converti en une séquence respective de mots numériques de n bits par un codage PCM (ou MIC, Modulation par Impulsions et Codage).

5. Un procédé selon la revendication 1, dans lequel les bits de poids fort des mots numériques de n bits correspondent aux bits de poids fort des composantes réelles et imaginaires des symboles QAM.

6. Un procédé selon la revendication 1, dans lequel ledit au moins un signal vocal est associé à au moins une voie téléphonique respective sous une forme analogique ou numérique.

7. Un procédé selon la revendication 1, dans lequel plusieurs porteuses de ladite partie de porteuses attribuée, peuvent être attribuées de nouveau pour acheminer des données quand, suite à une analyse d'une signalisation téléphonique, les voies téléphoniques respectives sont identifiées comme étant silencieuses.

8. Un émetteur pour l'utilisation dans un système de transmission d'un signal vocal et des données numériques vers une paire de câbles d'abonné au téléphone, lié à une source de données numériques à travers un port d'interface numérique, le système étant fourni avec une unité de signal de ligne à multitonalité discrète , désignée ci-après sous le nom de DMT, produisant un signal DMT constitué d'une pluralité de porteuses ayant des fréquences différentes et utilisant des premiers moyens de mise en séquence de tonalité pour attribuer lesdites données numériques aux porteuses du signal DMT, et utilisant un premier codeur de constellation connecté à un processeur de transformation de Fourier discrète inverse, désignée ci-après sous le nom d'IDFT, pour moduler les porteuses dudit signal DMT attribuées à la transmission de données, l'émetteur comprenant :

- au moins un port d'interface vocale connecté à une source de signal vocal extérieure correspondante,
- un circuit convertisseur respectif connecté à chaque port d'interface vocale pour convertir ledit signal vocal en au moins une séquence de mots numériques de n bits,
- des seconds moyens de mise en séquence de tonalité connectés à chacun des circuits convertisseurs pour attribuer une partie desdites porteuses dudit signal DMT à une transmission vocale,
- un convertisseur numérique analogique couplé au dit processeur d'IDFT pour convertir le signal DMT en un signal analogique pour l'envoyer sur ladite paire de câbles;

caractérisé par :

- un circuit de synchronisation connecté à chacun des circuits convertisseurs et au dit processeur d'IDFT pour synchroniser lesdits mots numériques de n bits avec lesdites trames dudit signal de ligne DMT;
- un second codeur de constellation recevant ladite au moins une séquence de mots numériques de n bits et étant connecté au dit processeur d'IDFT, ledit second codeur de constellation étant configuré pour moduler ladite partie desdites porteuses attribuées à la transmission vocale avec au moins une séquence de symboles QAM de n bits qui correspondent à ladite au moins une séquence de mots numériques de n bits, où chaque symbole QAM a une composante réelle constituée des bits impairs desdits mots numériques de n bits et une composante imaginaire constituée des bits pairs desdits mots numériques de n bits; et
- un chargeur de bits configuré pour acheminer ladite au moins une séquence de symboles QAM en synchronisation avec ladite séquence de mots numériques de n bits sur les porteuses respectives attribuées à la transmission vocale.

9. Un émetteur selon la revendication 8, dans lequel les bits de poids fort des mots numériques de n bits correspondent aux bits de poids fort des composantes réelles et imaginaires des symboles QAM.

10. Un émetteur selon la revendication 8, dans lequel ladite au moins une source vocale extérieure est au moins une voie téléphonique analogique et ledit au moins un circuit convertisseur est au moins un codeur de modulation par impulsion et codage, désignée ci-après sous le nom de PCM, et dans lequel au moins un port d'interface vocale est connecté au dit au moins un codeur PCM respectivement, pour transformer ledit au moins un signal vocal de ladite au moins une voie téléphonique en au moins une séquence correspondante de mots PCM de n bits, et dans lequel ledit au moins un codeur PCM est connecté auxdits seconds moyens de mise en séquence de tonalité, et dans lequel ledit au moins un codeur PCM et ledit processeur d'IDFT sont connectés à un synchroniseur pour synchroniser lesdites trames DMT avec ladite au moins une séquence de mots numériques de n bits dudit codeur PCM.

11. Un émetteur selon la revendication 8, dans lequel ledit au moins un signal vocal arrive à travers au moins une voie téléphonique extérieure connectée au dit au moins un port d'interface vocale.

12. Un émetteur selon la revendication 8, dans lequel un processeur pour analyser la voix au dit au moins un port d'interface vocale, est connecté à un allocateur de porteuses, en lui envoyant des instructions pour attribuer de nouveau, à la transmission de données, des porteuses attribuées autrefois à la transmission vocale lors de l'identification d'une voie vocale correspondante comme étant silencieuse.

13. Un émetteur selon la revendication 8, dans lequel ladite paire de câbles d'abonné de téléphone est une paire de câbles torsadés.

14. Un émetteur selon la revendication 10, dans lequel une pluralité de ports d'interface vocale sont connectés à un nombre correspondant de codeurs PCM, et dans lequel un concentrateur PCM est connecté auxdits codeurs PCM et auxdits seconds moyens de mise en séquence de tonalité.

15. Un émetteur selon la revendication 11, dans lequel un processeur pour analyser les signaux de commande téléphoniques à ladite voie téléphonique, est connecté à un allocateur de porteuses, en envoyant des instructions pour attribuer de nouveau, à la transmission de données, des porteuses attribuées autrefois à la transmission vocale lors de l'identification d'une voie téléphonique correspondante comme étant silencieuse.

16. Un émetteur selon la revendication 11, dans lequel ledit au moins un port d'interface vocale est un port d'interface vocale PCM connecté à un matériel de voies téléphoniques PCM extérieur d'un poste téléphonique, auxdits seconds moyens de mise en séquence de tonalité et à un synchroniseur connecté au dit processeur d'IDFT pour synchroniser lesdites trames DMT avec ladite au moins une séquence de mots numériques de n bits dudit matériel de voies téléphoniques PCM extérieur.

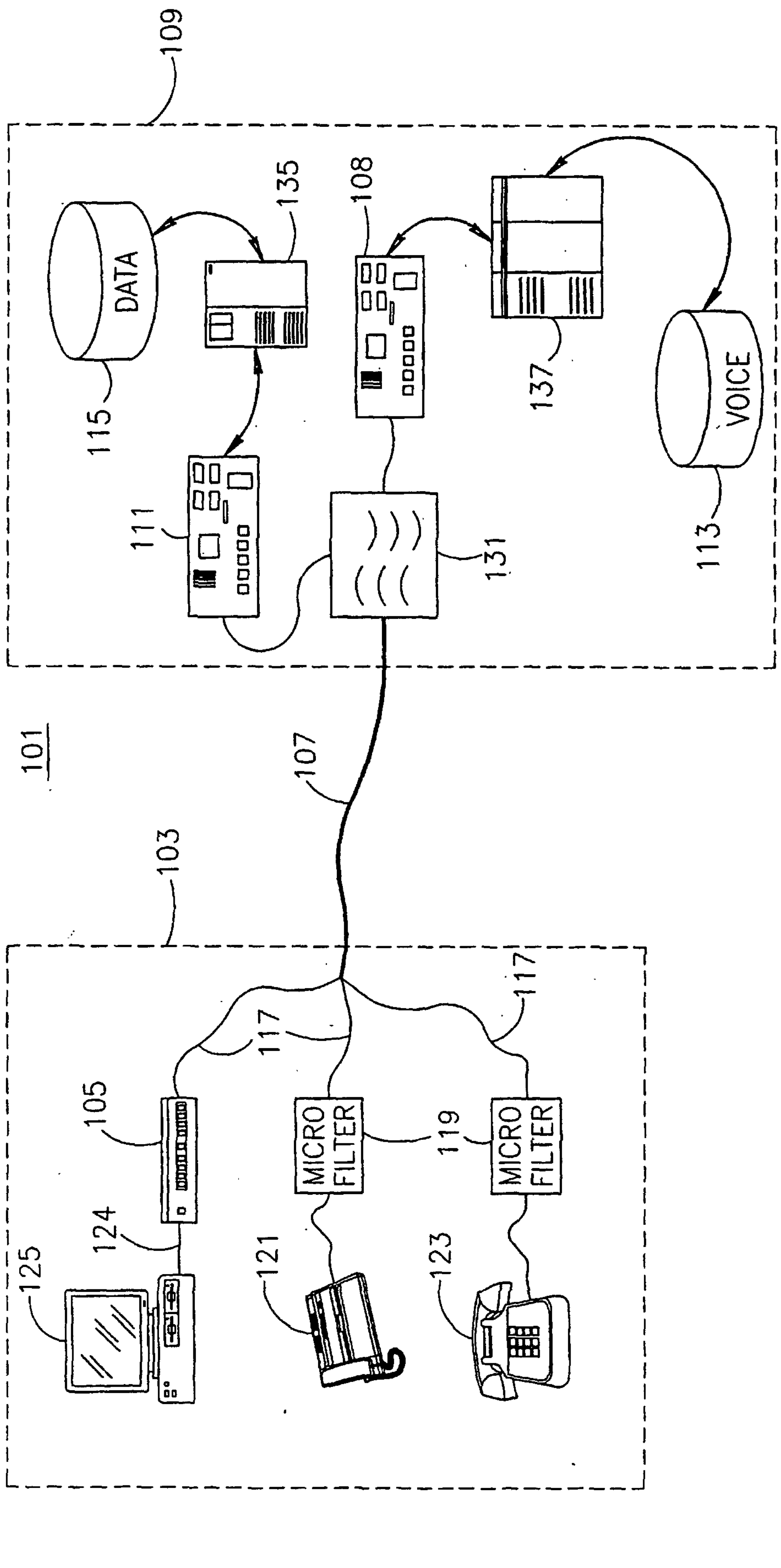


FIG.1
PRIOR ART

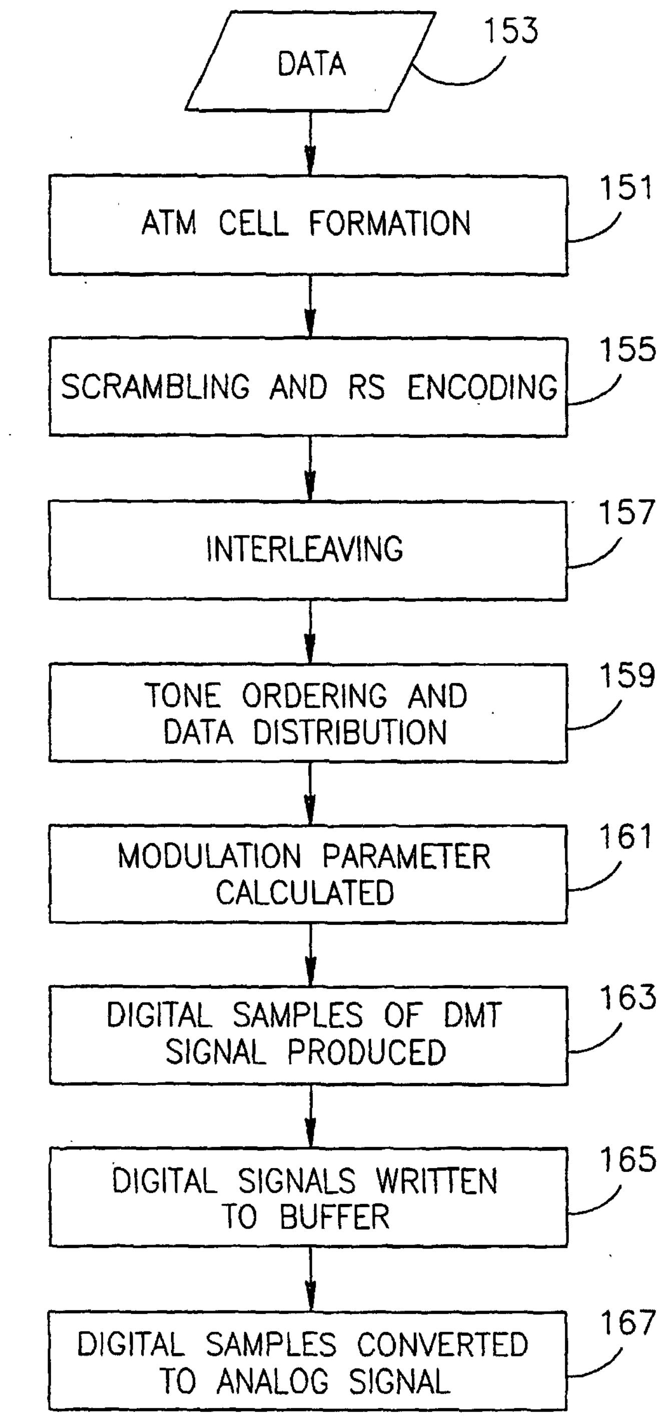


FIG.2
PRIOR ART

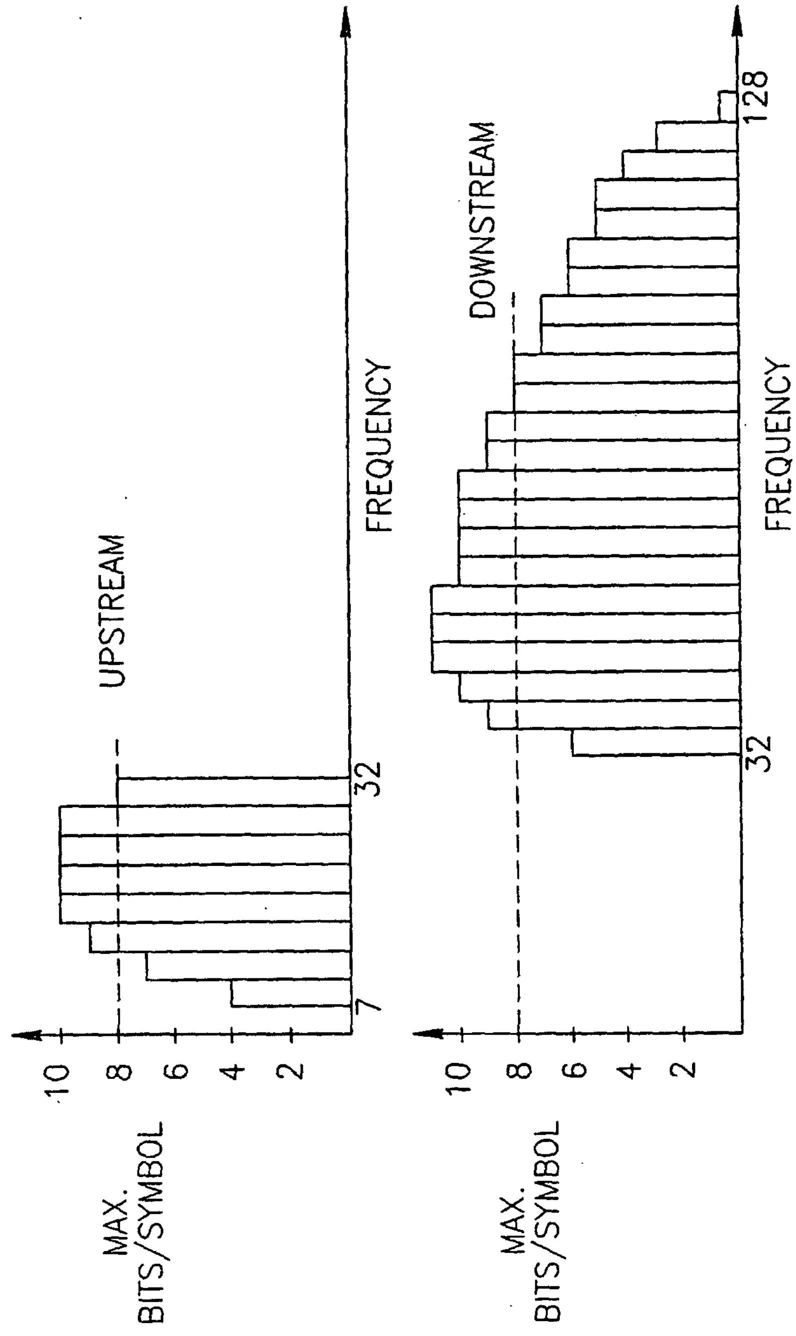


FIG.3
PRIOR ART

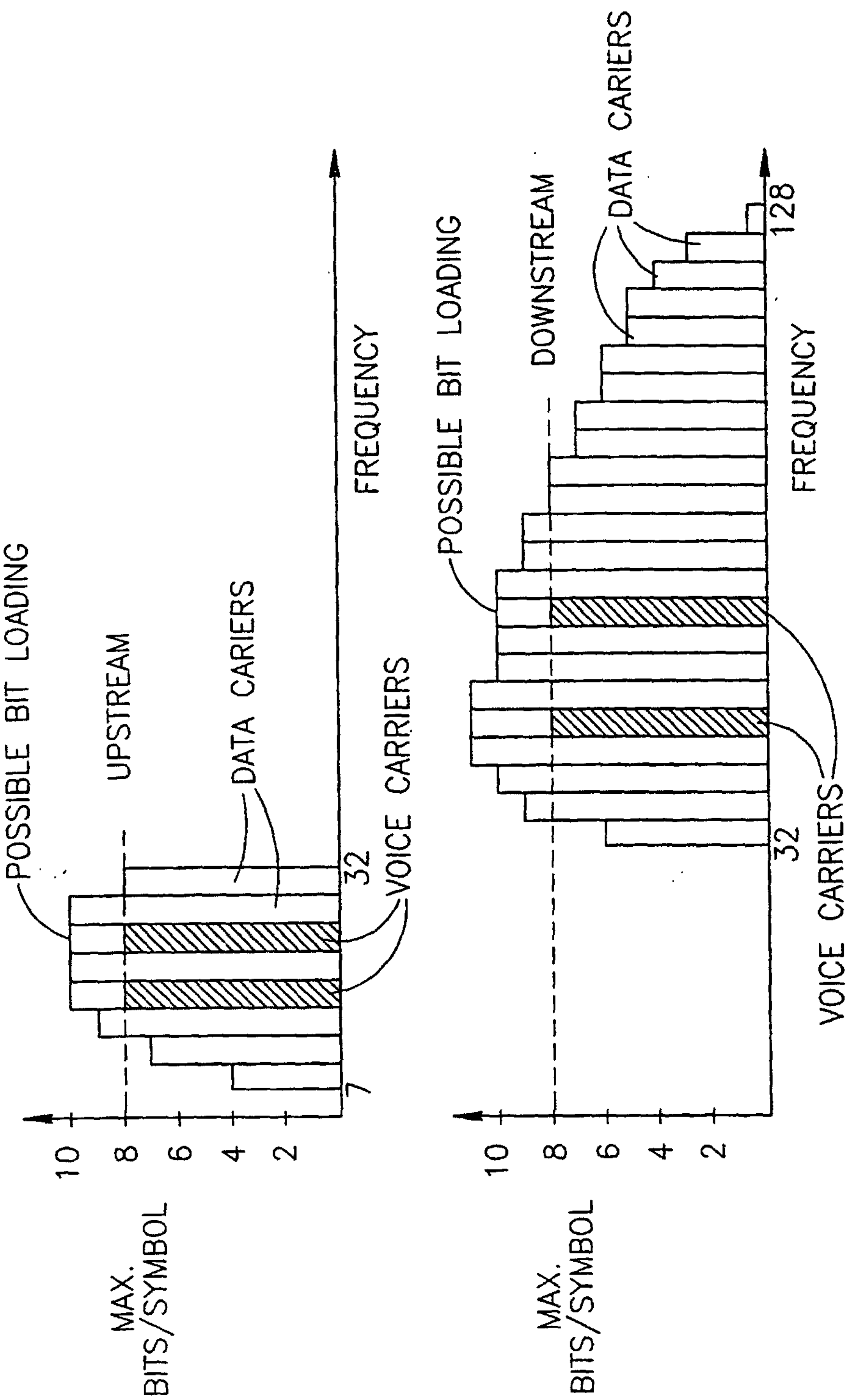
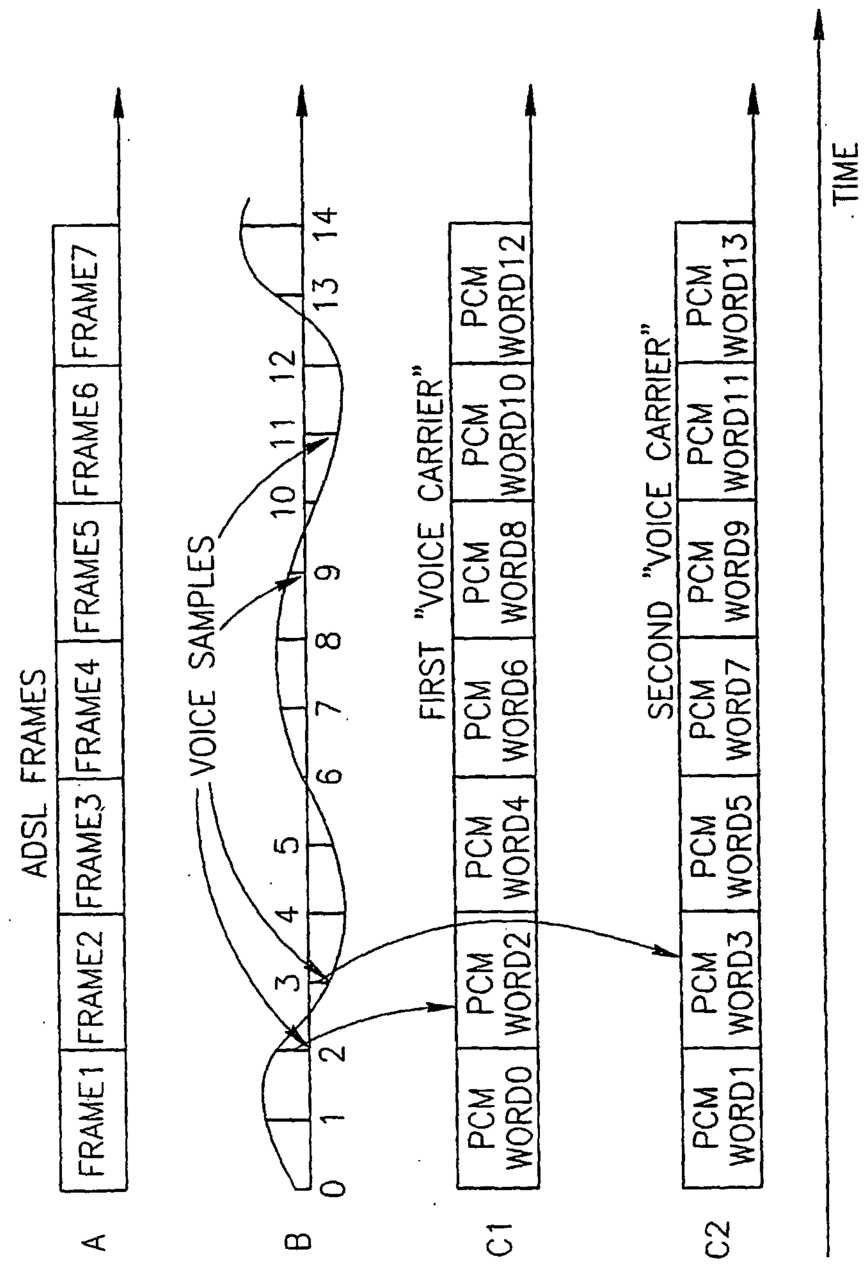

MAX. BITS/SYMBOL
10
8
6
4
2
7
32
POSSIBLE BIT LOADING
UPSTREAM
DATA CARIERS
VOICE CARRIERS
FREQUENCY
MAX. BITS/SYMBOL
10
8
6
4
2
32
128
POSSIBLE BIT LOADING
DOWNSTREAM
DATA CARIERS
VOICE CARRIERS
FREQUENCY

FIG.4



A
ADSL FRAMES
FRAME1
FRAME2
FRAME3
FRAME4
FRAME5
FRAME6
FRAME7
B
VOICE SAMPLES
0
1
2
3
4
5
6
7
8
9
10
11
12
13
14
C1
FIRST "VOICE CARRIER"
PCM WORD0
PCM WORD2
PCM WORD4
PCM WORD6
PCM WORD8
PCM WORD10
PCM WORD12
C2
SECOND "VOICE CARRIER"
PCM WORD1
PCM WORD3
PCM WORD5
PCM WORD7
PCM WORD9
PCM WORD11
PCM WORD13
TIME

FIG.5

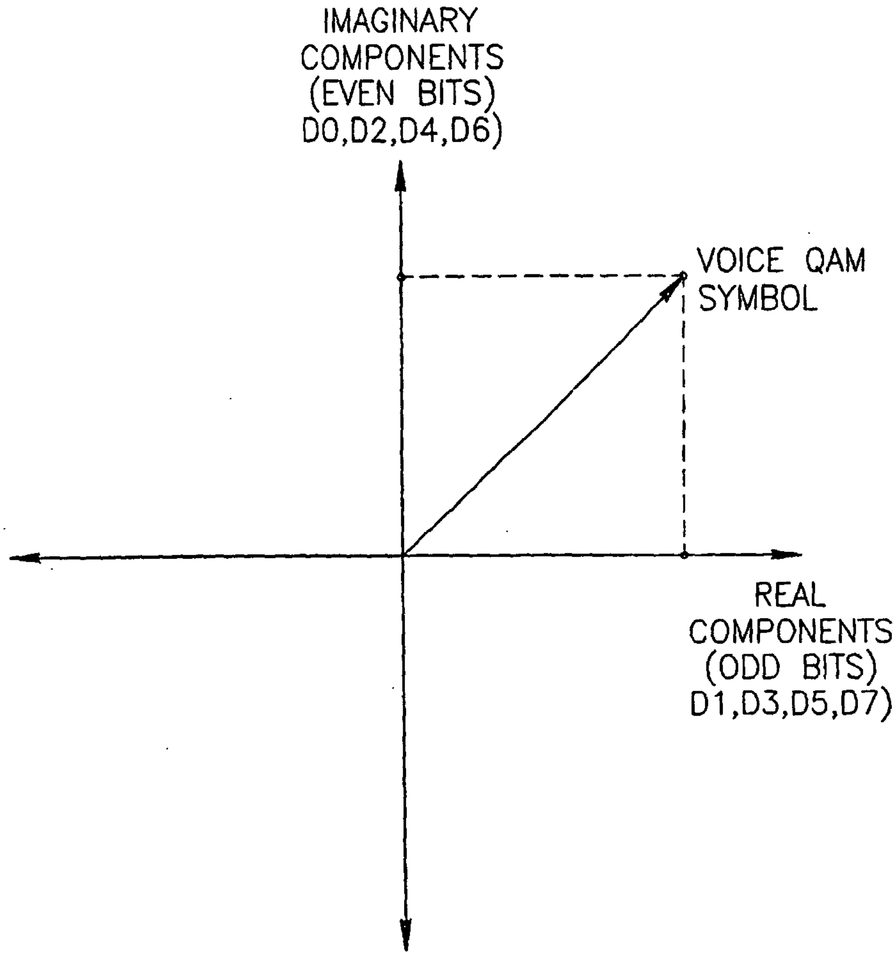

IMAGINARY
COMPONENTS
(EVEN BITS)
D0,D2,D4,D6)
VOICE QAM
SYMBOL
REAL
COMPONENTS
(ODD BITS)
D1,D3,D5,D7)

FIG.6A

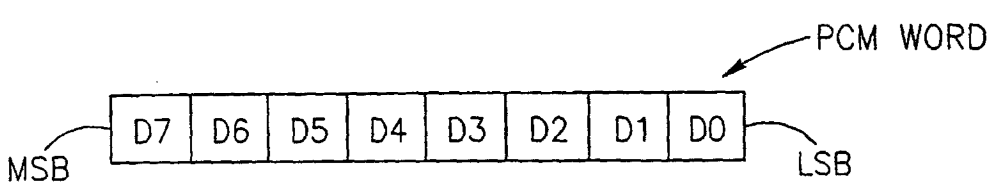

PCM WORD
MSB
D7
D6
D5
D4
D3
D2
D1
D0
LSB

FIG.6B

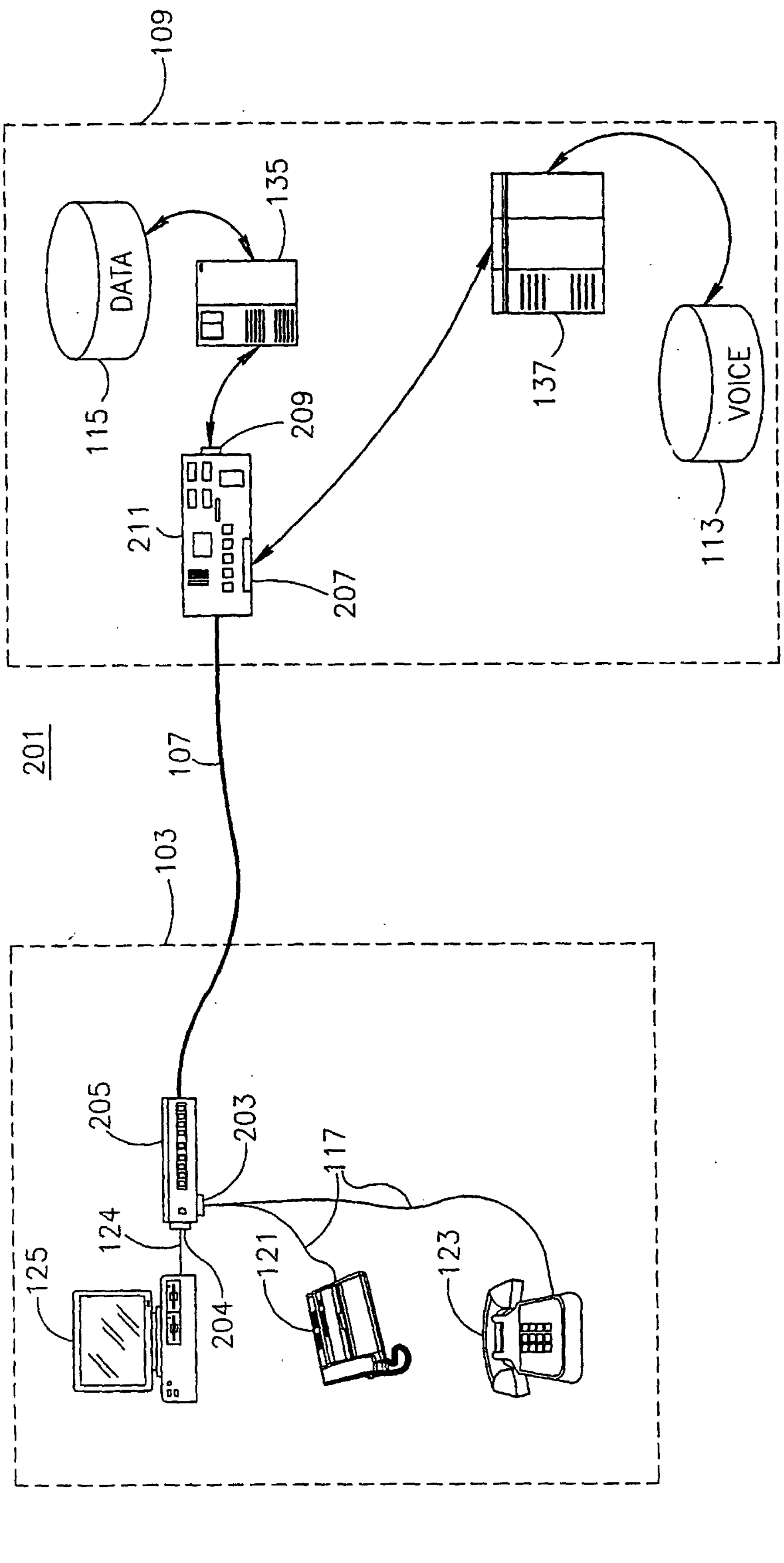

109
DATA
135
115
209
211
207
137
VOICE
113
201
107
103
205
203
117
124
125
121
123
204

FIG.7

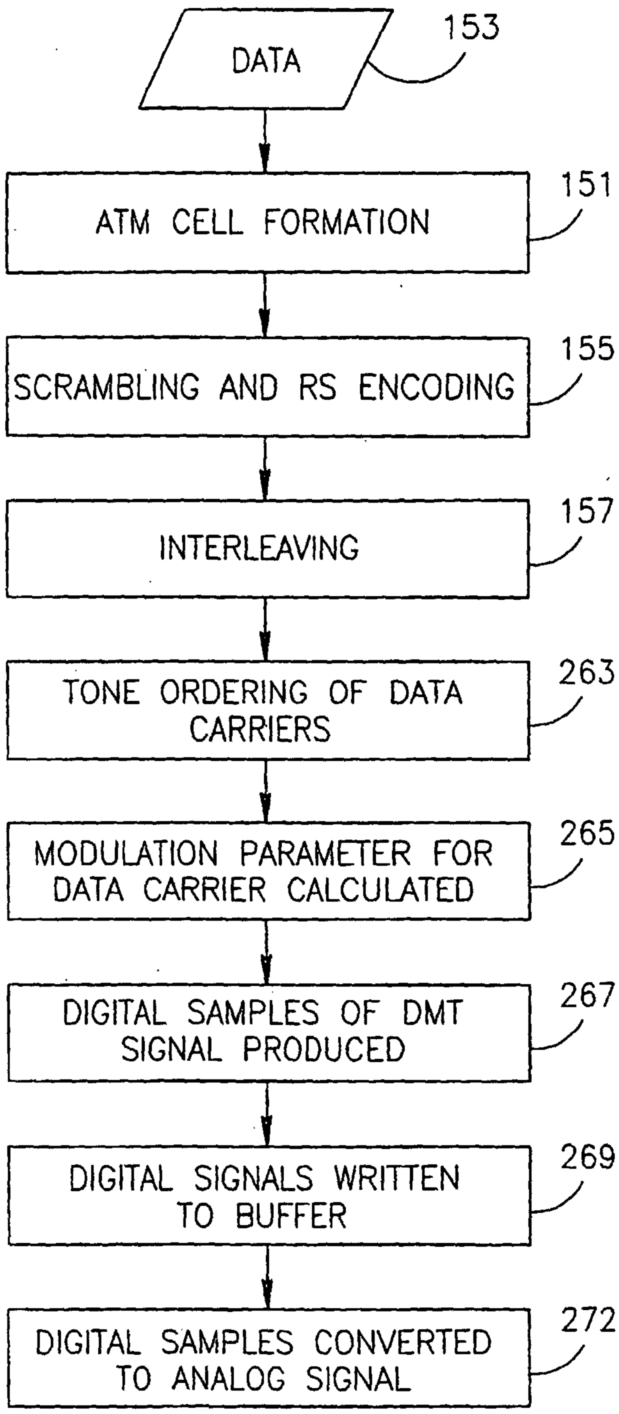

DATA
153
ATM CELL FORMATION
151
SCRAMBLING AND RS ENCODING
155
INTERLEAVING
157
TONE ORDERING OF DATA CARRIERS
263
MODULATION PARAMETER FOR DATA CARRIER CALCULATED
265
DIGITAL SAMPLES OF DMT SIGNAL PRODUCED
267
DIGITAL SIGNALS WRITTEN TO BUFFER
269
DIGITAL SAMPLES CONVERTED TO ANALOG SIGNAL
272

FIG.8A

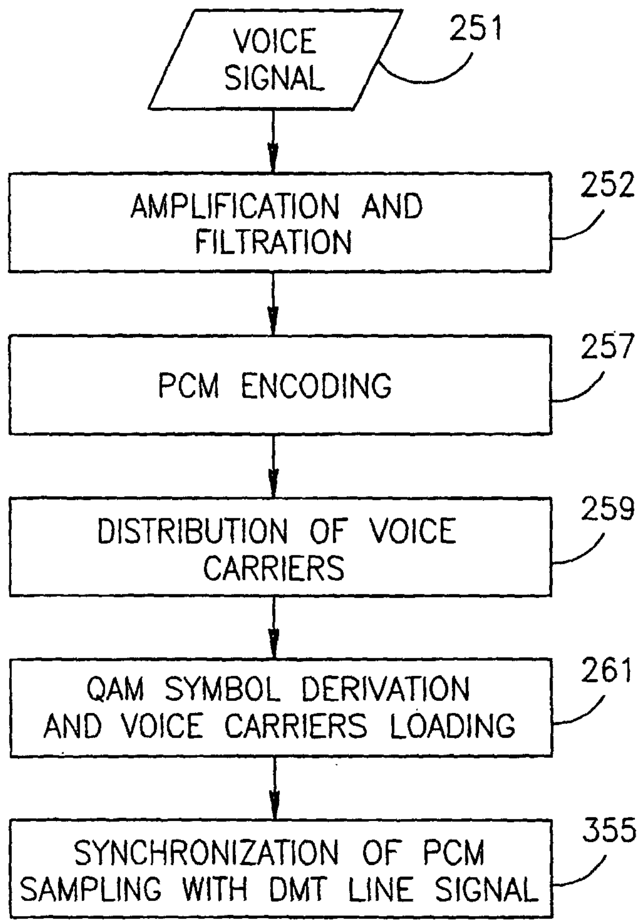

VOICE SIGNAL
251
AMPLIFICATION AND FILTRATION
252
PCM ENCODING
257
DISTRIBUTION OF VOICE CARRIERS
259
QAM SYMBOL DERIVATION AND VOICE CARRIERS LOADING
261
SYNCHRONIZATION OF PCM SAMPLING WITH DMT LINE SIGNAL
355

FIG.8B

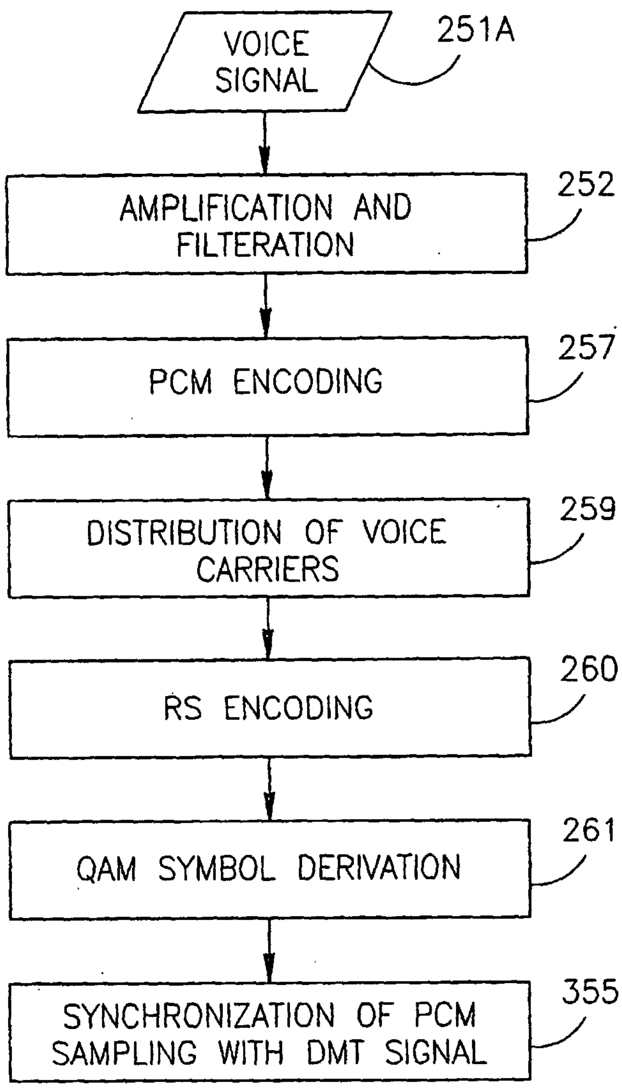

VOICE SIGNAL
251A
AMPLIFICATION AND FILTERATION
252
PCM ENCODING
257
DISTRIBUTION OF VOICE CARRIERS
259
RS ENCODING
260
QAM SYMBOL DERIVATION
261
SYNCHRONIZATION OF PCM SAMPLING WITH DMT SIGNAL
355

FIG.9

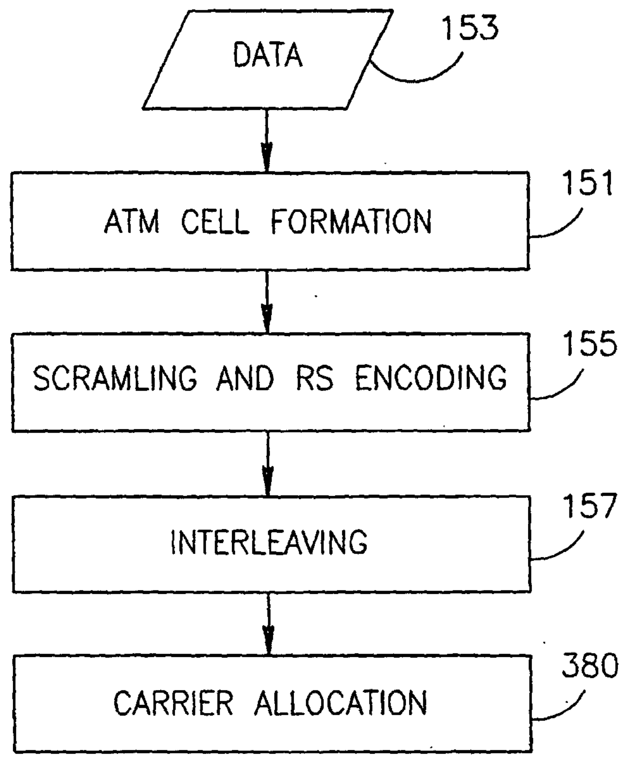

DATA
153
ATM CELL FORMATION
151
SCRAMLING AND RS ENCODING
155
INTERLEAVING
157
CARRIER ALLOCATION
380

FIG.10A

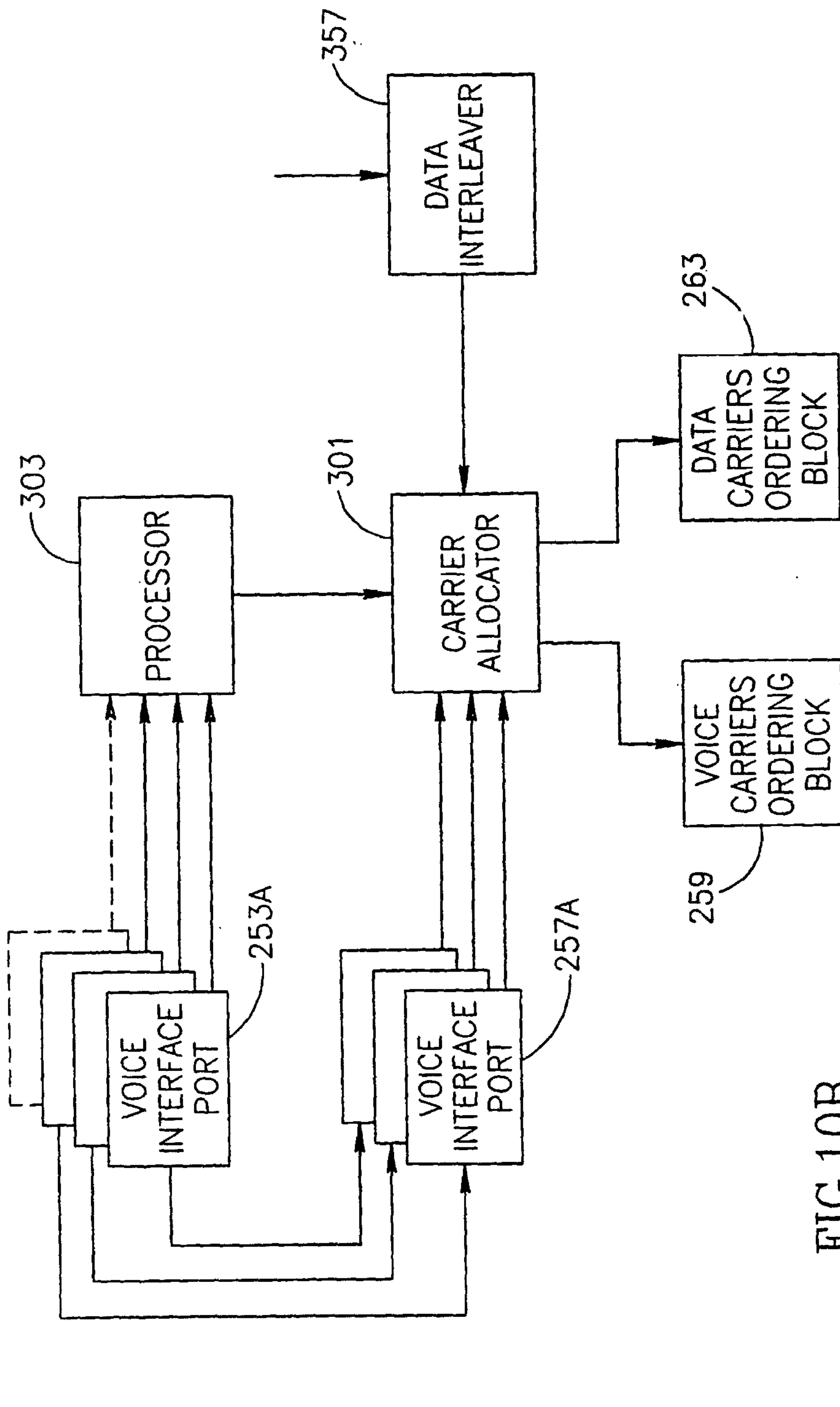

357
DATA INTERLEAVER
303
PROCESSOR
301
CARRIER ALLOCATOR
263
DATA CARRIERS ORDERING BLOCK
VOICE CARRIERS ORDERING BLOCK
259
VOICE INTERFACE PORT
253A
VOICE INTERFACE PORT
257A

FIG.10B

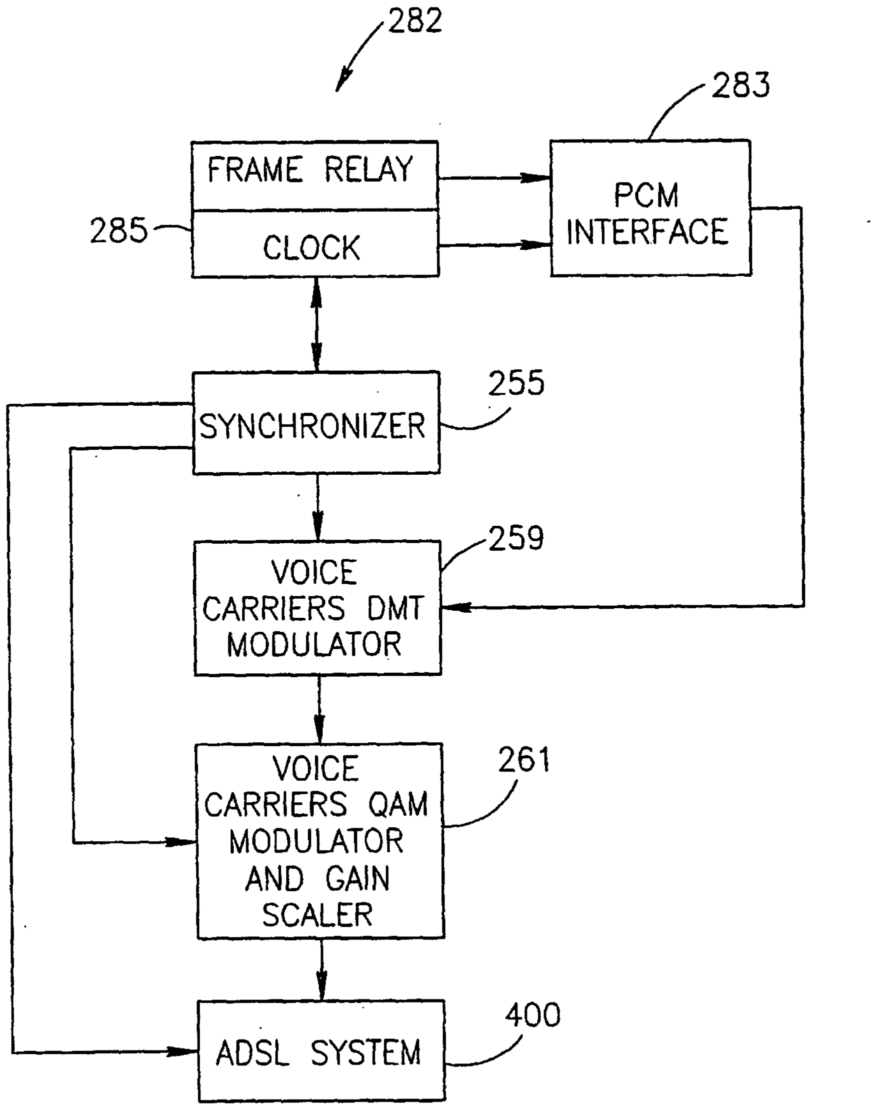

282
283
FRAME RELAY
CLOCK
285
PCM INTERFACE
SYNCHRONIZER
255
VOICE CARRIERS DMT MODULATOR
259
VOICE CARRIERS QAM MODULATOR AND GAIN SCALER
261
ADSL SYSTEM
400

FIG.11

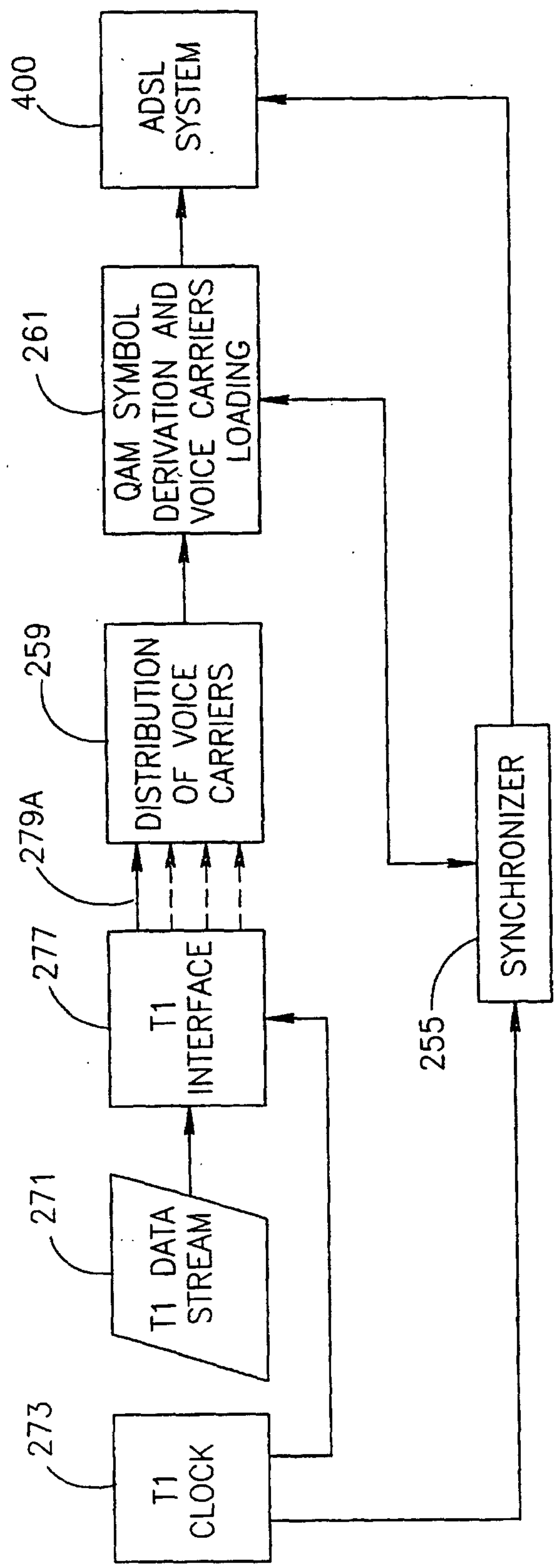

400
ADSL SYSTEM
261
QAM SYMBOL DERIVATION AND VOICE CARRIERS LOADING
259
DISTRIBUTION OF VOICE CARRIERS
279A
277
T1 INTERFACE
271
T1 DATA STREAM
273
T1 CLOCK
255
SYNCHRONIZER

FIG.12

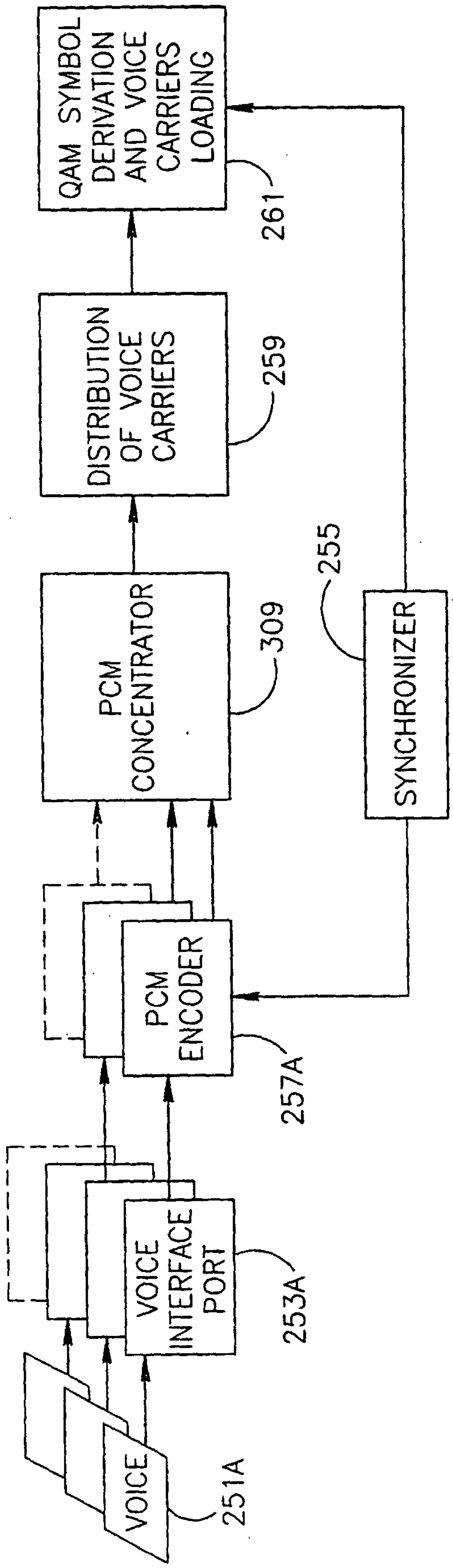

QAM SYMBOL DERIVATION AND VOICE CARRIERS LOADING
261
DISTRIBUTION OF VOICE CARRIERS
259
PCM CONCENTRATOR
309
SYNCHRONIZER
255
PCM ENCODER
257A
VOICE INTERFACE PORT
253A
VOICE
251A

FIG.13