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(57) Abstract

Apparatus and method for detecting periods of voice signal silence and allowing a data station to acquire the voice signal TDM time slot to be used for data transmission. This is enabled since all data stations on the multipoint communication line listen to the line during the first part of a voice channel time slot. If silence is detected during a voice signal time slot, all the data stations on the communication line know that the remainder of the voice time slot can be used for data transmission.

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APPARATUS AND METHOD FOR REALLOCATING TIME DIVISION MULTIPLEXING VOICE BANDWIDTH FOR DATA TRANSMISSION

CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of U.S. Provisional Patent Application Serial. No. 60/079,350, filed on March 25, 1997, and entitled "Techniques for Dynamic Reallocation of Time Division Multiplexing Voice Bandwidth for Data Transmission," which is incorporated by reference herein in its entirety.

BACKGROUND OF THE INVENTION

FIELD OF THE INVENTION

The present invention generally relates to an apparatus and method for enabling a plurality telephone services that can be utilized simultaneously on a single telephone line. More particularly, voice time slot bandwidth recovery using time division multiplexing for derived POTS. This derived POTS utilizes a methodology of simultaneous voice data transmission capability wherein the data can be either voice digital type data.

DESCRIPTION OF THE RELATED ART

Presently, telephone companies can offer only one set of services to any and all POTS-type devices on each wire pair at the premise, because current POTS service requires one (1) line per service. This is because device types are mutually exclusive, and consequently only one device type can utilize the service line at any one time. A further limitation exists for the telephones, such that all extensions are connected to the same conversation. Presently, if multiple services are desired, an additional line is required for each additional service. This is most evident in situations such as a second loop for a fax machine or a “teen line” to separate parent telephone calls from those of children in a household. There are added costs for each additional line.

Also, telephone companies today cannot command any additional service revenue from the usage of extra phones, modems, and fax operations on a single line. Until now, telephone companies could not offer any extra beneficial service to the premise. Accordingly, there is a need to develop an apparatus and method to transmit a plurality of data signals in parallel with multiple analog plain old telephone services (POTS) signals, thereby providing multiple plain old telephone type services on a single telephone line.

With such an apparatus and method for enabling simultaneous multiple telephone-type services on a single telephone line, the telephone companies can offer numerous sets of services to any and/or all POTS-type devices on each wire pair at the premise. The apparatus and method for enabling simultaneous multiple telephone type services on a single telephone line is disclosed by the modem described in the commonly assigned and copending U.S. Patent Application entitled, "Apparatus and Method for Transmission of Voice Band Signals Over a DSL Line," serial number XX/XXX,XXX, filed on March 25, 1999, herein incorporated by reference. The modem apparatus and method for transmission of voice band signals over a DSL line allocates bandwidth to voice signals to provide a derived POTS, which enables multiple telephone type services on a single line. However, the allocation of bandwidth to voice signal significantly decreases the bandwidth available for simultaneous data transmission, especially over long loops where less than the total bandwidth is available.

SUMMARY OF THE INVENTION

To achieve the advantages and novel features, the present invention is generally directed to a data communications apparatus and method that allows an user to utilize, simultaneously, multiple telephone-type services to any and/or all POTS-type devices on each wire pair at the premise. More particularly, the present invention provides voice time slot bandwidth recovery using time division multiplexing for derived POTS.

One embodiment of the present invention provides an apparatus and method for detecting periods of voice signal silence and allowing a data station to acquire the voice signal TDM time slot to be used for data transmission. This is enabled since all data stations on the multipoint communication line listen to the line during the first part of a voice channel time slot. If silence is detected during a voice signal time slot, all the data stations on the communication line know that the remainder of the voice time slot can be used for data transmission. Whichever data station currently has access to the line, and there can only be one, begins or resumes transmitting data during the remainder of the voice time slot. This allows the bandwidth allocated for the voice signal available for data transmission when the voice signal is silent.

Further, another embodiment of the present invention accomplishes this by using a time division multiplexing (TDM) method. This method utilizes a form of multiplexing in which transmission time is broken into segments, wherein each segment carries one element of one signal for each device that is separately addressable.

The modem apparatus used in the present invention includes a memory containing a plurality of program routine sequences and a processor that performs the selected program routine sequences to enable simultaneous multiple access techniques. Suitable modems for this purpose include the modem described in commonly assigned and co-pending U.S. Patent Application entitled "APPARATUS AND METHOD FOR COMMUNICATING VOICE AND DATA BETWEEN A CUSTOMER PREMISES AND A CENTRAL OFFICE", Serial Number 08/962,796 filed on, November 3, 1997, herein incorporated by reference, and the modem described in commonly assigned and co-pending U.S. Patent Application entitled "APPARATUS AND METHOD FOR A MULTIPOINT DSL MODEM", Serial Number 09/031,226 filed on, February 26, 1997, herein incorporated by reference.

BRIEF DESCRIPTION OF THE DRAWINGS

The accompanying drawings incorporated in and forming a part of the specification illustrate several aspects of the present invention, and together with the description, serve to explain the principles of the invention. In the drawings:

FIG. 1 is a view of the central office (CO) wire centers and user premises layout of the prior art.

FIG. 2 is a view of the CO wire centers and user premises layout of the present invention, with many of the multiple telephone-type services depicted.

FIG. 3A is a block diagram of the CO POTS interface and modem apparatuses of FIG. 2.

FIG. 3B is a block diagram of the user premises POTS interface and modem apparatuses of FIG. 2.

FIG. 4 is a block diagram of the digital signal processor engine of FIGS.. 3A and 3B.

FIG. 5 is a block diagram of the derived POTS circuitry for the digital signal processor of FIG. 4.

FIG. 6 is a block diagram of the packet using the multipoint protocol that provides allows each device to be separately addressable.

FIG. 7 is a diagram of a representative frequency spectrum utilized by the modems of FIGs. 3A, 3B, 4 and 5 to transmit the multipoint protocol packets of FIG. 6.

FIG. 8 is a diagram of a representative Frequency spectrum utilized by the modems of FIGs 3A, 3B, 4 and 5 to transmit the multipoint protocol packets of FIG. 5, when the frequency band normally utilized by POTS devices is not in currently in use.

FIGs.. 9A through 9C show comparisons of usage of line capacity using the digital methods of voice transmission for different total line capacities.

FIG. 9A shows the usage of line capacity the digital method of voice transmission for a total line capacity of 768 kbps per second with a symbol rate of 64 kHz and 12 bits per symbol.

FIG. 9B shows the usage of line capacity using the digital method of voice transmission for a total line capacity of 512 kbps per second with a symbol rate of 64 kHz and 8 bits per symbol.

FIG. 9C shows the usage of line capacity using the digital method of voice transmission for a total line capacity of 384 kbps per second with a symbol rate of 64 kHz and 6 bits per symbol.

FIGs. 10A through 10E show voice and data slot alignment within a time division multiplexing frame, including the bandwidth recovery method of the present invention.

FIG. 10A shows a time division multiplexing format for two derived POTS channels and a data channel.

FIG. 10B shows a more efficient use of the time division multiplexing format for the two derived POTS channels and data channel of the present invention.

FIG. 10C is a block diagram representative of a time division multiplexing frame after dropping the utilization of the derived POTS slot for a second station.

Illustrated in FIG. 10D is a block diagram of the time division multiplexing frame has been relocated to an upstream time slot for the remaining channel to increase efficiency of data transmission for the remaining data device.

FIG. 10E is a block diagram representative of the time division multiplexing frame format wherein a downstream voice slot is silent and is made available for data transmission.

Reference will now be made in detail to the description of the invention as illustrated in the drawings. While the invention will be described in connection with these drawings, there is no intent to limit it to the embodiment or embodiments disclosed therein. On the contrary, the intent is to cover all alternatives, modifications, and equivalents included within the spirit and scope of the invention as defined by the appended claims.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring now in detail to the drawings in which the reference numerals indicate like parts throughout several views.

FIG. 1 illustrates the plain old telephone system (POTS) networks including data communication modems 45 of the prior art.

The POTS network includes numerous user premises 41, wherein each user premises is connected to a central office wire center 11, via a subscriber line 27. Each subscriber line 27 is connected to the user premises 41, which further connects to a user premises line 47, for distribution of POTS service throughout the user premises. Usually, there are numerous POTS devices connected to each user premises line 47, such as telephones 44, fax machines 42, personal computers (PCs) 46, and the like. It is also known, (but not shown), that it is possible to have multiple subscriber lines 27 connected to each user premises, thereby creating two separate user premises lines 47 within each user premises.

As noted previously, each user premises is connected, via a subscriber line 27, to a central office wire center 11. The subscriber line 27 is connected to a POTS switch 19 that separates the analog POTS signals from data signals. The POTS signals are sent from the POTS switch 19 to the other central office wire centers, via the public switch telephone network (PSTN) 22. Modem data signals can be separated from the POTS analog signals at POTS switch 19, and are connected to network equipment 16, for connection with digital data networks such as, for example, the Internet 24.

A brief discussion of an example of the signals generated in the applied system environment of the prior art from the user premises and transmitted through the central office wire center, via either the PSTN or Internet networks, and back to a user premises, will now be detailed.

When a user wishes to place a telephone call on device 44A, for example, the user picks up the receiver and puts the subscriber line 27 in an off-hook condition that is detected at the central office wire center 11, by closed switch hooks (not shown). The off-hook condition signals the central office wire center 11, via subscriber line 27, to accept an outgoing call by allowing a flow of D.C. current and a dial tone of 480 Hz to be sent to device 44A. The outgoing telephone call signals are transmitted, as described previously, via subscriber line 27 to POTS switch 19. The analog POTS system signals are separated from the modem signals, and the POTS signals are transmitted, via the PSTN 22, to the destination central office wire center 11 of the destination user premises 41. The analog signal is further directed towards a POTS switch 19 within the destination central office wire center 11. The signal is transmitted, via subscriber line 27, to the destination user premises 41. The analog signal enters the destination user premises 41, via subscriber line 27, and is connected to the user premises line 47 that distributes the signal to be received throughout the destination user premises 41. This is the path in which a POTS call is transmitted.

Now, a description of digital signals to/from the user premises will be described. When a user desires to transmit digital data over a network via his personal PC 46 or the like, the digital signals from the digital device are transformed into analog signals, via multiplexing by modem 45. The signals are transmitted over the user premises line 47 to the subscriber line 27 for final delivery to the local central office wire center 11. The digitally multiplexed analog signals going into POTS switch 19, can be separated from the analog voice POTS signals, and directed to network equipment 16, for further transmission of the data signals over the Internet 24, or they can be transmitted to the destination CO wire center 11 over the PSTN 22. The digital data signals sent via the Internet 24 are received at the destination central office wire center 11 by the network access service equipment 16. The signals are transmitted to the POTS switch 19 and over destination subscriber line 27 to the destination user premises 41. The multiplexed signals are received at the user premises line 47, for distribution to all equipment connected to the user premises distribution line. The multiplexed signals are demultiplexed, within the destination modem 45, and converted back into a digital signals, which are transmitted to the digital device 46 connected to the modem.

FIG. 2 illustrates the plain old telephone system (POTS) networks including data communication modems (14 and 50) of the preferred embodiment. The data communication modems include the aforementioned apparatus and methods for enabling the simultaneous multiple telephone type services on a single line. FIG. 2 illustrates that a variety of services may be connected at the CO wire center 11 in accordance with the present invention. These services may include those requiring digital communications such as Internet access or home management and security services and those requiring analog communications such as voice communication or any service that normally communicates via a standard POTS connection. Again, the operation of such services are generally understood and a further discussion of them is not necessary in order to describe the operation of the present invention. As further illustrated in FIG. 2, the POTS voice devices, telephone 44 and standard fax machine 42, establish communications on the frequency band between 0 kHz and about 4 kHz.

A second transmission frequency band is defined at a higher frequency level than the POTS frequency band and is used in the transmission of digital subscriber line (DSL) communications that provides multiple access techniques of the preferred embodiment. The DSL modems 50 provide both the physical layer and higher layer functions needed to provide the simultaneous multiple access.

The different equipment devices at the user premises can be identified and accessed by a multiple access code (MAC) address as determined by the modem 50, or by the assigned available frequency range within the bandwidth of the communication. Now, the different types of services will be described with regard to FIG. 2.

Derived POTS phone 43A is an ordinary POTS phone that is connected to modem 50 instead of directly to subscriber line 47 as in the case of normal POTS service. Modem 50 encodes the analog signal received from phone 43A as necessary and transmits the encoded audio at an representative average data rate of 8000 samples per second and performs a reverse function in the received direction. This transmission uses a frequency band outside of the normal POTS frequency band of 0 kHz to 4 kHz and does not interfere with normal POTS services utilized by standard POTS devices, such as facsimile machine 44B and POTS phones 44A that are connected directly to subscriber line 27. Thus, the Derived POTS phone 43A acts to the user as a separate telephone line that may also communicate over the PSTN network.

The PC 46 may transmit and receive data via modem 50 from the Internet or from a local area network (LAN) or other point-to-point type of data transmissions network.

The home security and power meter reading system device 58 provides monitoring and controlling of various home functions such as a security system. It also provides the ability for communicating home functions data to a local utility such as gas usage, electricity usage, water usage, and the like.

All the unique service devices connected to modems 50, as shown and described with regard to FIG. 2, can be accessed via unique addresses. For each particular Telco service provided, that service provides the user a unique address for each new service premise device. Thus, those and only those unique service devices are enabled.

Each of the additional service devices illustrated in FIG. 2 are connected via modems 50 to the user premise line 47. This user premise line is further connected to one subscriber loop 27 that connects to the CO wire center 11. The signals from each of the service devices are modulated via modem 50 and input to the CO wire center plain old telephone system ("POTS") splitter 12 which separates the POTS communications that are now transmitted in the frequency band between 0 kHz and 4 kHz. These POTS signals are identified in POTS splitter 12 and separated from the multiple service signals operating at a higher frequency at POTS splitter 12. The POTS voice signals are separated from the data signals and transmitted to POTS switch 19 for communications over the PSTN 22 or other networks (not shown). The data signals and Internet data signals are separated from the voice POTS signals in POTS splitter 12 and forwarded on the master modem 14. Master modem 14 uses one of the aforementioned multiplexing technique to separate the representations of the signals of the derived POTS devices from those for the digital devices. The derived POTS signals are forwarded to POTS switch 19 on communication line 17 where they appear as signals no different from those normally received via standard POTS lines. Master modem also forwards the digital signals from the digital devices at customer premises 41 on communication line 15 for further transmission through the NAS equipment devices 16 to the Internet 24 and other LAN networks on communication line 23.

The Derived POTS phones 43 each may have a standard telephone number or may share a number with the other devices. The Derived POTS phone 43 would have standard Telco POTS features and billing. The Derived POTS phone 43 operation is enabled by the modem apparatus of the present invention herein described with regard to FIG. 3B through FIG. 12.

FIG. 3A is a block diagram of the CO wire center multichannel data communications device modem (modem 14) constructed in accordance with the present invention. The typical configuration of the central wire office 11 multichannel data communication device 14 is connected, via a POTS splitter 12, to the subscriber line 27. The analog signals output from POTS splitter 12 into the central office multichannel data communications device 14, are connected through communication links into the POTS interface 13. The central office multichannel data communications device 14 provides for multiple analog lines to be input and converted to digital signals, due to the efficiency of the processor 35 within the central office multichannel data communication device 14. Because multiple analog input lines are permitted, device 14 may require multiples of the analog POTS interface hardware 32, dial access arrangement (DAA) logic 33 and analog front end (AFE) logic 34.

The analog POTS interface hardware 32 connects analog signal line to the dial access arrangement (DAA) logic 33. The dial access arrangement (DAA) logic 33 provides surge protection and impedance matching. A line protection circuit (not shown) protects the multichannel communications device 14 against line surges, lightning strikes, and the like. The line protection circuit is then further connected to an impedance and isolation circuit (not shown), via a communication link. The Impedance and Isolation circuit also contains circuitry to detect Ring Indicator on and Off-hook conditions.

The impedance and isolation circuit is comprised of an impedance matching circuit (not shown) before being connected to the two-to-four wire hybrid interface (not shown). The dial access arrangement (DAA) logic 33 connects the analog signals to the analog front end (AFE) logic 34, via a communication link.

The analog front end (AFE) logic 34 converts the analog signal to a digital data signal. The AFE 34 is connected to a communication link which is connected to a receiver (not shown). The receiver receives the analog signals and converts the analog signal by using an analog-to-digital converter (not shown). A driver (not shown) drives the signals across a communication link to the impedance and isolation circuit (not shown) of DAA 33, after receiving signals from the driver's digital-to-analog converter (not shown). The receiver analog-to-digital converter and driver digital-to-analog converter are both connected to the bi-directional digital communication link. Ring indicator and off-hook conditions are processed in a Ring Indicator (RI) Off-Hook (OH) Impedance Controller (not shown).

The AFE logic 34 transmits the digital signal to the DSP logic 35 for reconstruction of the digital data. Multiple analog front end logic components 34 may be connected to a single DSP, CPU, ASIC or other processor logic 35, due to the high processing speed of such processor logic.

In alternative embodiments of the invention, the multiple dial access arrangements (DAA) logic components 33 and analog front end logic components 34 are not necessary to practice the present invention, and may be omitted in some applications where the dial access arrangement (DAA) logic 33 and analog front end logic 34 are shared between numerous analog POTS interface hardware 32, as will be understood by those skilled in the art.

DSP logic 35 reconstructs the digital signal streams into usable digital data by stripping error control information, data compression and the like added by the far-end modem. The reconstructed digital data is transmitted from the DSP logic 35 through the host interface 36 to the host DTE 16 devices for further transmission over the Internet 24, or other like digital networks.

DSP logic 35 also reconstructs the samples of the analog signals received at customer premises 41 from POTS devices 43 that are connected to modems 50. These are then forwarded to POTS switch 19 via either multiple POTS analog interfaces 37A-X or POTS pulse code modulation (PCM) interface 38. Analog interfaces 37A-X transmit analog signals in the frequency band from 0-4 kHz that are identical to those normally transmitted by a standard POTS device connected directly to subscriber line 47. POTS PCM interface 38 transmits standard μ-Law or A-Law digital representations of multiple analog channels as a multiplexed stream of 8-bit samples using standard formats such as that prescribed by Bell Communications Research (Bellcore) Generic Requirements GR-303-CORE "Integrated Digital Loop Carrier System Generic Requirements, Objectives, and Interface". In either case, POTS switch 19 receives these signals in a form that is no different from standard POTS circuitry and provides a separate and independent access to the PSTN 22 from each derived POTS device at customer premises 41.

Likewise, DSP logic 35 receives digital data from host digital device 16 via connection 15 and host interface 36. It also receives signals representing analog POTS channels from POTS switch 19 via connections 17 and either POTS analog interfaces 37A-X or POTS PCM interface 38. Analog interfaces 37A-X receive analog signals in the frequency band from 0-4 kHz that are identical to those normally transmitted by POTS switch 19 via a direct connection to a standard subscriber line. POTS PCM interface 38 receives standard μ-Law or A-Law digital representations of multiple analog channels as a multiplexed stream of 8-bit samples using standard formats such as that prescribed by prescribed by Bell Communications Research (Bellcore) Generic Requirements GR-303-CORE "Integrated Digital Loop Carrier System Generic Requirements, Objectives, and Interface". In either case POTS switch 19 provides a separate and independent access from the PSTN to each derived POTS device at customer premises 41.

FIG. 3B is a block diagram of the single POTS line multichannel data communication device (modem 50) constructed in accordance with the present invention. The multichannel data communication device (modem 50), is substantially similar to the CO wire center multichannel data communication device 13, defined in Fig 3A., except that device 50 is configured to accept multiple POTS device connections in addition to one or more digital device connection.

In the typical configuration, the user premises line 47 is connected to POTS line interface 62. The POTS line interface 62 is connected to Dial Access Arrangement interface 63, Analog Front End 64, digital signal processor logic 65, and the device communications interface 66, as described in 3A above as items 3X. The digital signal processor logic 65 is connected to the host digital device or data terminal equipment (DTE), by a digital device communications interface 66 bus via a communication line, which connects to a device such as a digital fax, digital phone, personal computer (PC), or the like. The digital signal processor logic 65 is also connected to the multiple analog device interface 67 via a communication line. The multiple analog device interface allows the communication device 50 to connect multiple analog devices for derived POTS service of the present invention. These analog POTS devices for derived POTS service typically are telephones, however, other devices that normally operate via a direct connection to a subscriber line such as facsimile machines, audio services, power meter reading, home management, security systems and the like are also contemplated by the inventor.

Communications device 50 can be for example but not limited to, a data service unit (DSU), modem, or any other communication device capable of frame relay communication. In the preferred embodiment, communication device 50 is a DSU, which contains proprietary address determination logic. Central office location 11 is typically the local telephone companies' local exchange office which connects via copper wire pair 27 to a remote customer location 41, which can be for example a residential or business location.

As shown in FIG. 4, the digital communication link 72 is connected to the digital signal processor engine (35 or 65, herein referred to as 65), which includes a digital signal processor (DSP) or application specific integrated circuit (ASIC) chip 71, which is connected to read only memory (ROM) 77 and random access memory (RAM) 74. ROM 77 can be comprised of either regular ROM or RAM memory, flash memories, erasable programmable read only memory (EPROMs), electrically erasable programmable read only memory (EEPROMs), or other suitable program storage memories. RAM memory 74 can be comprised of static or dynamic RAM, EEPROM, or other suitable data storage memories.

The transmission sequence routines for controlling the transmission format (*i.e.* multipoint protocol for control of data transmission and time division multiplexing for control of voice versus data transmission) are normally within the digital signal processor engine (65) program ROM 77. Transmission sequence routines can be downloaded from digital devices, usually a PC connected to the DTE interface 66 (FIG. 3B), into the digital signal processor engine 65 program RAM 74 program area 75. It is contemplated by the inventor that an updated routine may be downloaded to the modem apparatus to update the address determination routines.

The derived POTS logic 80 of the present invention can be implemented in hardware, software, firmware, or a combination thereof. In the preferred embodiment(s), the derived POTS logic 80 is implemented in software or firmware that is stored in a memory and that is executed by a suitable instruction execution system.

The address determination routines are also in the digital signal processor engine (65) program ROM 77. Address determination routines can likewise be downloaded from digital devices into the digital signal processor engine 65 program RAM 75 program area, so that an updated address determination routines may be downloaded to the modem apparatus.

The incoming signals on digital line 72 are input into the DSP engine 71 for processing. Control signals and digital input/output signals are communicated across digital communication link 73. Digital communication links 72 and 73 can be comprised of 8, 16, 32, 64, 168 or other bit sized digital parallel communication links. Communication links 72 and 73 can also be comprised of bit serial or other types of chip-to-chip signal communication links. The DSP or ASIC 71 of the digital signal processor engine 65 is connected, via communication link 73 interface 36 or 66 as illustrated in FIGs.. 3A and 3B.

Referring to FIG. 5, which is a functional block diagram representative of the derived POTS logic 80, as shown in FIG. 4. As previously noted, the derived POTS logic 80 residing within the DSP or ASICs 71 is connected to the multiple analog device interface 67 and the digital device communication interface 66. Outgoing derived POTS signals are received by the data communication device 50 by the multiple analog interface 67 as described above. The analog signal is transmitted via bus 73A to the voice band analog-to-digital converter logic 81.

The voice band analog-to-digital converters convert each analog signal to a digital signal by sampling the analog signal at a representative sample rate of 8 kHz, for example, to a produce a stream of digital samples. This analog-to-digital conversion can be done using either linear conversion, or Mu or A law conversions. It is contemplated by the inventor that multiple voice band analog-to-digital converters 81A-X can be utilized to support any number of analog source signals to be transmitted.

Next, the voice samples for each analog source signal are accumulated in voice sample buffers 82A-X so that a block of samples for each analog source signal is available to be sent over the subscriber line 27 as a burst. The size of the block has a major impact on both the delay experienced by the voice band signal and the efficiency at which the voice signal can be transmitted since there is some fixed amount of overhead associated with each burst. The analog samples are grouped into frames and applied to the modulating signal in bursts at a rate that is much higher than the rate at which the pairs of samples were generated. In order to do this it is of course necessary to collect blocks of samples in the voice sample buffers 82A-X until the point in time arrives at which the pairs of analog samples can be applied to the modulating signal in a burst.

Multiplexer 89 selects from one of the buffers of voice samples as the current source for following processing blocks. Typically all analog-to-digital converters 81A-X generate samples of their analog input signals continuously and at the same rate and therefore all produce the same size blocks of samples in the voice sample buffers 82A-X at the same rate. In this case, multiplexer 89 can operate in a simple circular manner in which each voice sample buffer is selected at the input in turn until a complete block has been transmitted. When the contents of the last buffer have been transmitted, the multiplexer returns to the first buffer as the source. More complex schemes to control multiplexer 89 can be readily applied to accommodate cases where the sample rates of the A-to-D converters 81A-X and or the sizes of the voice sample blocks are not the same for all analog input sources 67.

The logic of block 80 allows for a single modem to alternately transmit a signal representing one or more voice channels or signals representing data from a digital device 46 connected to digital device communication interface 66 via multipoint protocol control block 93. Multiplexer 87 alternately selects blocks of data for transmission comprised of either voice samples from multiplexer 89or digital data from multipoint protocol control block 93. The scheme for controlling multiplexer 87 derives from the technique used for managing bandwidth on the subscriber line among multiple signal sources. Since time-division multiplexing is used to manage use of the line between voice data from various sources and digital data, all voice sources are assigned segments of time within a periodically repeating and constant length time interval. Multiplexer 87 is controlled by logic (not shown) that is clocked at the symbol rate and generates a gating signal that corresponds to the time slots assigned to each transmitted voice channel.

Multipoint protocol control block 93 controls usage of the portion of the time available for transmission of data using the commonly understood and widely employed multipoint polling protocol technique. The following refers to the operation of block 93 at master modem 14 (FIG. 2). This block transmits data received via digital device interface 66 using a packet format similar to that shown in FIG. 6. These packets are sent to multiplexer 87 during time slots made available for data transmission. Block 93 can also transmit a poll to one customer premises modem 50 inviting that modem to transmit and temporarily relinquishing control of the line to that modem. Block 93 receives upsteam data sent in reponse to polls via de-multiplexer 92 and transfers the information portion of these responses to digital device interface 66.

The following refers to the operation of block 93 at customer premises modem 50. Block 93 receives downstream messages from de-multiplexer 92. If the address in the message indicates the message is for this station, block 93 forwards any information contents to digital device interface 66. If the received message is addressed to this station and contains a poll, block 93 transmits data received via digital device interface 66 to mulitplexer 87 using a packet format similar to that shown in FIG. 6.

The significant feature of this arrangement is that multipoint protocol block 93 receives data from de-multiplexer 92 and transmits data to multiplexer 87 only during the time slots allocated for data transmission. It is not necessary for multipoint protocol block 93 to format protocol packets to fit within the data time slots. Transmission of a protocol frame can begin and end anywhere within a data time slot as will be illustrated in later discussion.

The serial data stream selected by multiplexer 89 is scrambled and mapped into a stream of two-dimensional symbols in blocks 85 and 86 using the widely known techniques commonly applied to transmission of digital data using passband modulations. It is contemplated by the inventor that the passband modulation transmitter 88 may use Carrier-less Amplitude-Phase (CAP) Modulation, Quadrature Amplitude Modulation (QAM) or other known modulation capable of transmitting a stream of 2-dimensional samples. CAP, which is a widely known technique for data transmission over a Digital Subscriber Line (DSL) is very similar to QAM. The main distinctions are that CAP omits the multiplication of the in-phase and quadrature components of the transmitted symbol by the cosine and sine of the accumulated carrier phase at the transmitter and receiver and replaces the baseband filters applied to these components in QAM with orthogonal (Hilbert) passband filters. Other bandpass modulations suitable for use on a digital subscriber line may be used as well.

An incoming signal is received by the analog front end and transmitted to a passband modulation receiver block 91 of design as appropriate for the transmitted modulation that recovers the stream of 2-D symbols transmitted from another station. Receiver 91 forwards the stream of 2-D symbols to the symbol decoding block 98 and descramber and symbol degrouping block 99 using the technique appropriate for data transmitted using the widely known techniques commonly applied to transmission of digital data using passband modulations.

De-multiplexer 92 alternately delivers blocks of received data comprised of digitized voice samples to de-multiplexer 95 or comprised of digital data to multipoint protocol control block 93. The scheme for controlling de-multiplexer 92 derives from the technique used for managing bandwidth on the subscriber line among multiple signal sources. Since time-division multiplexing is used to manage use of the line, de-multiplexer 92 is controlled by logic that is clocked at the symbol rate and generates a gating signal that corresponds to the time slots assigned to each received voice channel.

. The digital signal produced by multipoint protocol control block 93 by extracting the information field from the protocol packets is then transmitted across bus 73B to the digital device communication interface 66 for further transmission to the connected digital device 46, or the like.

During time slot assigned to voice channels used by this modem, blocks of voice samples are delivered by de-multiplexer 92 to de-multiplexer block 95 which selects one of the receiver voice sample buffers 96A-X to hold the block of samples. The de-multiplexer 95 is controlled using an appropriate scheme to match the way in which multiplexer 89 at the transmitter selected blocks of voice samples for transmission. Blocks of voice samples are recovered from the received signal at a rate that is much higher than the continuous rate at which they must be delivered to digital-to-analog converters 97A-X for accurate reproduction of the original analog signal. Voice sample buffers 96A-X are therefore required to match the instantaneous rates of the input and output.

The voice samples are then delivered from the voice sample buffers 96A-X to the voiceband digital-to-analog converters 97A-X which change the samples back into their analog form and transmit the signals to the multiple analog interface device 97 across bus 73A.

With reference now to Fig. 6, shown is a schematic view illustrating a communications packet 101 that may be transported by the modem 50 of Figs. 3A and 3B. Packet 101 can be similar in format to a standard frame relay communication packet. It contains a beginning flag 103 followed by a fixed length address and control field to identify both the sender and the intended recipient of the message and possibly the type of message being sent. It also contains a variable length information or payload section followed by a cyclical redundancy check (CRC) error detection code used to ensure the integrity of the transported information. Finally, frame 106 contains the one octet end flag used to signal the end of the packet.

Packets of a format similar to that shown in FIG. 6 are used to transmit data between digital DTE devices across the network.

Turning now to the drawings, FIG. 7 is a diagram illustrating various frequency bands used for communications over a digital subscriber line. As is known in the prior art, (POTS communications are transmitted in the frequency band 111 defined between about 0 (DC) and about 4 kHz. A second transmission frequency band 112 is defined at a higher frequency level than the POTS frequency band 111, and is used in the transmission of digital subscriber line (DSL) communications. A guard dead band 113 is typically provided to separate the two transmission frequency bands 111 and 112.

The DSL transmission frequency band 112 is more broadly denominated as "xDSL", wherein the "x" generically denominates any of a number of transmission techniques within the DSL family. For example, ADSL – Asymmetric Digital Subscriber Line, RADSL -- Rate Adaptive Digital Subscriber Line, HDSL -- High-Bit-Rate DSL, *etc*. As is known, xDSL transmission frequency bands may encompass a bandwidth of greater than 1 MHz. As a result, and for the reasons described above, without the addition of extra equipment such as POTS filters, splitters, etc. xDSL signals are not compatible with attached POTS type equipment, such as telephones, PSTN modems, facsimile machines, *etc.*

In accordance with one aspect of the invention, a multichannel data communication device (modem 50) is provided for achieving efficient data communications between a customer premises 41 and a central office 11 across a local loop 27, by dynamically allocating a transmission frequency bandwidth for transmitting data. Certainly, one of the factors motivating the development of the present invention is the expanded demand for higher speed communications in recent years. This enhanced demand is primarily attributed to communications over the Internet.

The present invention dynamically allocates a data transmission frequency band in response to POTS communications across the same line. More particularly, the present invention may utilize the frequency band otherwise allocated for POTS/voice transmission, at times when there is no present demand for transmitting voice information as illustrated in FIG. 8. When, however, there is a demand for voice transmissions on the POTS line, then the present invention reallocates the transmission frequency band for the data communications so that there is no overlap or interference with the POTS transmission frequency band 111, and so that there is not significant interference to POTS type attached equipment.

FIGs. 9A-9C show three examples intended to illustrate use of the technique of the current invention for transmission of analog voice signals across a digital subscriber line by simply converting the stream of samples representing the analog signal to a stream of bits and transmits this across the line in the same manner as any digital data stream. Typically each voice sample is produced by the analog-to-digital converted at a rate of 8 kHz and is represented in μ–Law or A-law format using 8 bits producing a digital stream of 64 kbps. It is assumed that assumed that the symbol rate of the passband modulation is 64 kHz. It is also assumed that a single frequency band is used for both upstream and downstream transmission and that transmission in takes place in only one direction at time (i. e. it is "half duplex"). It is also assumed that time is broken into equal periodic intervals referred to as voice frames and that, during each of these time periods a block of voice samples is transmitted from each active (i. e. 'off hook') POTS device 43A-X connected to each addressable modem device 50 and, since POTS service normally provides full-duplex communications, one block of voice samples is also transmitted to each active POTS device.

In the examples of FIGs. 9A-9C it is assumed that there are currently two such devices however the example can easily be extended to apply to any number of devices provided there is sufficient bandwidth available to meet the data transmission needs.

The examples in FIGs. 9A-9C show the portion of the available bandwidth used for voice signal transmission and the resulting amount of bandwidth available for transmission of digital data and overhead with three different values of total available bandwidths equal to 384 kbps, 512 kbps and 768 kbps for FIGs. 9A-9C respectively.

For voice transmission as digital data, each voice channel requires 64kbps upstream and 64kbps downstream for a total bandwidth requirement of 256 kbps for two voice channels.

As shown in FIGs. 9A-9C, the portion of the voice frame time occupied by voice data increases as the available data rate decreases because the same amount of data must be sent during the voice frame time in every case. The remaining bandwidth available for data is just the total available bit rate minus the 256 kbps required for voice or 128 kbps, 256 kbps and 512 kbps for FIGs. 9A-9C respectively. This demonstrates the need to develop a method of the present invention to recover as much unused voice bandwidth as possible.

Referring now to FIG. 10A-10E, illustrated are the block diagrams representative of the derived POTS voice time slots and data time slot utilization of a transmission frame.

Referring now to FIG. 10A, illustrated is transmission sequences using Time Division Multiplexing for voice signals with two voice channels available, wherein both voice channels have signal activity. In this scheme, time is broken into regularly repeating periods referred to as transfer frame times 209.

Each active voice channel is assigned a consistent segment within this time during which to transmit the upstream signal (204, 205) and a second continuous segment to transmit the downstream signal (201, 202). The beginning and ending time of these segments relative to the beginning of the transfer frame time period are constant. Since the time segments for transmission are fixed, there is no need for a digital header preceding the voice samples. However, for timing purposes, there is a gap time between the voice and data segments. Any time remaining in the transfer frame period 209 is available for transmission of digital data (206). As this portion of the frame can potentially be used by any station to transmit digital data, use of this interval is managed by the master modem 14 (FIG. 2) using a polling protocol as described previously.

Illustrated in FIG. 10B is a block diagram representative of a technique to increase bandwidth efficiency by minimizing the number of gap time slots in the frame. In order for the derived POTS service to provide acceptable voice quality, the delay has to be kept below some reasonable value. Larger frame sizes have increased bandwidth efficiency, but voice quality requirements place an upper limit on the maximum frame size. For a given frame size, efficiency can be maximized by using a time division multiplexing frame format which minimizes the number of gap time slots. The gap time slot between the station one voice slot 211 and the station two voice slot 212 has been eliminated since both time slots originate from the same master modem 14 (FIG. 2). This elimination of the gap time slot between voice slot 211 and 212 greatly increases bandwidth efficiency by minimizing the number of gap time slots in the frame.

Another optimization technique is illustrated in FIG. 10C which is a block diagram representative of the time division multiplexing transfer frame utilizing dynamic activation. Since derived POTS channels are only activated when needed for telephone type service so that the bandwidth can be used for data when a derived POTS device is not in use. This requires voice time slots be allocated dynamically based on usage of the derived POTS line. During normal usage, the TDM format frame will have to change between frame formats providing zero, one, or more, derived POTS channels. As noted above, FIG. 10B illustrates the time division multiplexing frame format before dropping of the station to derived POTS device usage. Illustrated in FIG. 10C is the time division multiplexing frame format after dropping of the second derived POTS device usage. In FIG. 10C, the master modem 14 reassigns the now unused station 2 voice time slot to be used for data in slot 222. Also reassigned by the master modem 14 (FIG. 2) is the station 2 voice upstream time slot 214 which has been incorporated into the data time slot 224. While this may be easier to implement, it has the disadvantage of not eliminating a gap time slot which decreases bandwidth efficiency when using only one derived POTS voice channel.

Illustrated in FIG. 10D is the block diagram representative of the time division multiplexing frame format wherein the master modem 14 (FIG. 2) reconfigures the upstream station 1 voice time slot 223 (FIG. 10C) to the position adjacent to downstream station 1 voice time slot 231. This relocation of upstream station 1 voice time slot 223 (FIG. 10C) to the upstream station 1 voice time slot 232 greatly increases bandwidth efficiency by first removing a gap time slot, and second, allowing the data time slot 233 to be a larger contiguous data time slot.

It is clear that when using the time division multiplexing frame format which minimizes the number of gap time slots, changes in the position of the derived POTS time slots have been made both due to the changes in data rate and due to the changes in the number of active derived POTS channels. Both the changing of positions and the changing of duration of the time slots affects the actual delay in transmitting the derived POTS data through the communication system. This is a form of "delay jitter" with important characteristics that the limits of the jitter for every time division multiplexing frame format are known at design time. In order to run the derived POTS channel continuously with no time slips, this delay jitter has to be compensated for by buffering. In order that there be no timing slips, the receiver must always send the voice time slot back to the network at the same point in time regardless of changes in the inbound/outbound data rate and the number of derived POTS channels in use. This point in time should correspond to the latest time that the end of the time slot ever arrives for all time division multiplexing formats. When using time division multiplexing formats for which this point in time is earlier, buffering in the receiver should be used to hold this data until the fixed transmission point in time arrives. The amount of buffering is equal to the number of voice samples which are sent/received during the time interval equal to the difference between the earliest and latest arrival times of the end of the time slot. To eliminate timing slips and minimize delays when using voice time slots, the following two conditions should be met:

1. First, the transmit voice time slot boundary should coincide with the earliest time in which the end of the transmit voice time slot occurs for all the time division multiplexing formats for which it will be used.

2. The receive voice time slot boundary should coincide with the latest time at which the end of the receive voice time slot occurs for all the time division multiplexing formats which will be used.

Under these conditions, the additional delay imposed by the derived POTS channel for giving time division multiplexing formatting scheme for all line speeds and channel counts, is equal to the frame time plus the difference between the earliest and latest times at the end of a voice time slot occurs, plus the line transmission time. The higher the data transmission rate, (*i.e.*, shorter time slot), the addition buffering is all at the receiver end. At the lower data rate, the buffering is all at the transmitter. The jitter buffering system maintains a constant end-to-end delay by effectively moving buffered data between the transmitter and receiver as the time division multiplexing frame format changes.

Illustrated in FIG. 10E is a block diagram representative of the technique of the present invention that converts some or all of silent time in the voice time slots into bandwidth available for data time slot transmission. Since a large percentage of voice conversations consist of silence, if the bandwidth normally occupied by a voice derived POTS channel can be made available for data transmission during silent periods, the data bandwidth can be significantly increased. This is possible for any derived POTS channel which has periods of silence during the transmission session. This is illustrated in FIG. 10E when station 2 voice time slot 242 is silent. When a block of voice data has been assembled in voice sample buffers 82A-X (FIG. 5), prior to transmitting the block, it is analyzed to determine if the entire block represents silence. This can be done, for example, by determining the peak voice energy found in the block. If the buffer is determined to contain silence, rather than transmitting this across the line, the station does not transmit anything at all. It is important to note that transmission of voice silence, even if all the samples in the block are zero, does not result in silence being transmitting on the line. The binary data stream representing the block is scrambled in block 85 and encoded just as any data stream. The signal transmitted on the line to deliver a block of all-zero voice samples is the same in terms of signal energy as it is for transmitting any data. By transmitting true silence in lieu of a block of data representing silence, the station signals all other stations on the line that it will not use the time slot allocated to it for transmission of voice data. The station that would normally receive voice data from this station, upon detecting this true silence (i. e. no modualtion signal energy present) on the subscriber line , internally generates a block of voice samples representing silence throughout the entire block.

The one station that has access to the line may, after waiting a silence detection time 257, acquire the remainder of the time slot for data transmission, as shown as item 258. The silent detection time item 257 is required because it is not practical for all listening data stations to discriminate silence instantly. The data stations will have to listen for some short time period delay silence detection time 257 before concluding that the derived POTS time slot 242 is available for data transmission. This time period 257 represents a loss to the amount of time which can be recovered for data transmission from the unused derived POTS voice time slot 242.

The poll-response management of the multipoint data channel ensures that at any given time one, and only one, data station has access to the line for data transmission. This discipline is maintained independently of the status of the derived POTS channels. Therefore, if a voice or derived POTS station does not require the usage of its transmit derived POTS time slot because the input is silent, the derived POTS station can broadcast this condition to all other stations on the line by not transmitting any modulation energy on the line. Once this is done, the data station that currently has access to the line for data transmission can immediately begin transmitting in that derived POTS time slot. The data station which currently has access to the line for data transmission can transmit data until either the end of the derived POTS time slot or when the data device message end is reached. If the data message ends first, whatever other data station would normally transmit next uses the remainder of the derived POTS time slot just as it does for dedicated time slots.

The utilization of this time slot bandwidth recovery requires, for converting some or all of silent derived POTS time slots into bandwidth available for data transmission, the following. First, the reliable detection of a derived POTS session. Second, the reliable detection of silence during the derived POTS time slots. Third, communication of the availability of additional time for data transmission. This is due to the fact that the station that would normally use a time slot for derived POTS is not the same station which currently can use it to send data, therefore, there is a need to communicate the availability of the line bandwidth between the two stations.

Figure 10E also shows, for the purpose of illustration, an example of how the time slot of a silence voice source and the time slot made available for data may be used. In this example, when silence is detected during the time interval represented by block 257, tributary station 1 which has previously been invited by the master station to transmit and has previously transmitted some but not all of its protocol packet, now completes transmission of this packet during the first part of the newly available data time slot 258 as represented by block 259. Station 1 then relinquishes use of the line automatically. When the master modem receives the end of the response packet from station 1, it transmits a message containing a poll to station 2. This is started within the time remaining in data slot 258 as represented by block 260 and completed within the first part of the data slot 245 in block 261. The response to this poll by station 2 then begins also with data slot 245 at block 262 but does not complete.

The derived POTS logic 80, which comprises an ordered listing of executable instructions for implementing logical functions, can be embodied in any computer-readable medium for use by or in connection with an instruction execution system, apparatus, or device, such as a computer-based system, processor-containing system, or other system that can fetch the instructions from the instruction execution system, apparatus, or device and execute the instructions. In the context of this document, a "computer-readable medium" can be any means that can contain, store, communicate, propagate, or transport the program for use by or in connection with the instruction execution system, apparatus, or device. The computer readable medium can be, for example but not limited to, an electronic, magnetic, optical, electromagnetic, infrared, or semiconductor system, apparatus, device, or propagation medium. More specific examples (a nonexhaustive list) of the computer-readable medium would include the following: an electrical connection (electronic) having one or more wires, a portable computer diskette (magnetic), a random access memory (RAM) (magnetic), a read-only memory (ROM) (magnetic), an erasable programmable read-only memory (EPROM or Flash memory) (magnetic), an optical fiber (optical), and a portable compact disc read-only memory (CDROM) (optical). Note that the computer-readable medium could even be paper or another suitable medium upon which the program is printed, as the program can be electronically captured, via for instance optical scanning of the paper or other medium, then compiled, interpreted or otherwise processed in a suitable manner if necessary, and then stored in a computer memory.

The foregoing description has been presented for purposes of illustration and description. It is not intended to be exhaustive or to limit the invention to the precise forms disclosed. Obvious modifications or variations are possible in light of the above teachings. The embodiment or embodiments discussed were chosen and described to provide the best illustration of the principles of the invention and its practical application to thereby enable one of ordinary skill in the art to utilize the invention in various embodiments and with various modifications as are suited to the particular use contemplated. All such modifications and variations are within the scope of the invention as determined by the appended claims when interpreted in accordance with the breadth to which they are fairly and legally entitled.

What I claim is:

1. A bandwidth recovering data communications apparatus for supporting a plurality POTS communications devices and digital device using time division multiplexing, said data communications apparatus comprising:

interface circuitry connectable to a communications link;

a derived POTS interface circuitry connectable to a plurality of analog devices;

a digital interface circuitry connectable to a digital device;

silence recognition circuitry to detect when a signal representing silence from one of a plurality of analog devices is to be transmitted;

transmission circuitry that excludes transmission of a signal representing silence from one of a plurality of analog devices is to be transmitted; and

a logic circuitry that enables said data communications apparatus to transmit a signal from said digital device on said communications link, when said logic circuitry detects a silent time slot scheduled for use by one of said plurality of analog devices on said communications link.

2. The apparatus of claim 1, wherein said logic circuitry further comprises:

session detection circuitry to detect a derived POTS session on said communications line; silence detection circuitry to detect silence during a scheduled derived POTS time slot session on said communications line; and

bandwidth recovering transmission circuitry to enable transmission, on said communications link, of a signal from said digital device when said session detection circuitry detects a derived POTS session, and said silence detection circuitry detects silence during a scheduled derived POTS time slot.

3. The apparatus of claim 2, wherein said logic circuitry further comprises:

circuitry that determines if said data communications apparatus has current access to the communications line for enabling digital data transmission during a remainder of said silent time slot scheduled for use for derived POTS signals.

4. The apparatus of claim 2, wherein said logic circuitry further comprises:

dynamic activation time slot circuitry that activates a time slot for one of said analog devices when needed for service, so that transmission time can be maximized for said signal from said digital devices.

5. The apparatus of claim 2, wherein said logic circuitry further comprises:

relocation circuitry that relocates a transmission analog time slot to be adjacent to the closest analog time slot to greatly increases bandwidth efficiency by first removing a gap time slot, and allowing said digital device time slot to be a larger contiguous data time slot.

6. A method for use in a bandwidth recovering data communications apparatus for supporting a plurality POTS communications devices and digital device using time division multiplexing on a communication line, the method comprising the steps of: providing a derived POTS interface circuitry connectable to a plurality of analog devices; providing a digital interface circuitry connectable to a digital device;

detecting when a signal representing silence from one of a plurality of analog devices is to be transmitted; excludes transmission of a signal representing silence from one of a plurality of analog devices is to be transmitted; monitoring said communications line for a silent time slot scheduled for use by one of said plurality of analog devices on said communications link; determining if said data communications apparatus has current access to said communications line for enabling a signal for said digital device to be transmitted during a remainder of said silent time slot scheduled for use by one of said plurality of analog devices; and transmitting said signal for said digital device during a remainder of said silent time slot if said data communications apparatus has current access to said communications line.

7. The method of claim 6, further comprising the step of:

enabling transmission, on said communications link, of a signal from said digital device when detecting a analog session on said communications line, and detecting silence during a scheduled analog time slot session on said communications line.

8. The method of claim 6, further comprising the step of:

determining if said data communications apparatus has current access to the communications line for enabling digital data transmission during a remainder of said silent time slot scheduled for use for derived POTS signals.

9. The method of claim 6, further comprising the step of:

activating a time slot for one of said analog devices when needed for service, so that transmission time can be maximized for said signal from said digital devices.

10. The method of claim 9, further comprising the step of:

relocating a transmission analog time slot to be adjacent to a previous analog time slot to greatly increases bandwidth efficiency by first removing a gap time slot, and allowing said digital device time slot to be a larger contiguous data time slot.

11. A bandwidth recovering data communications apparatus for supporting a plurality POTS communications devices and digital device using time division multiplexing, said data communications apparatus comprising: a means for providing a derived POTS interface circuitry connectable to a plurality of analog devices;

a means for providing a digital interface circuitry connectable to a digital device; a means for detecting when a signal representing silence from one of a plurality of analog devices is to be transmitted; a means for excludes transmission of a signal representing silence from one of a plurality of analog devices is to be transmitted;

a means for monitoring a communications line for a silent time slot scheduled for use by one of said plurality of analog devices on said communications link;

a means for determining if said data communications apparatus has current access to said communications line for enabling a signal for said digital device to be transmitted during a remainder of said silent time slot scheduled for use by one of said plurality of analog devices; and a means for transmitting said signal for said digital device during a remainder of said silent time slot if said data communications apparatus has current access to said communications line.

12. The apparatus of claim 11, further comprising:

a means for enabling transmission, on said communications link, of a signal from said digital device when detecting a analog session on said communications line, and detecting silence during a scheduled analog time slot session on said communications line.

13. The apparatus of claim 11, further comprising:

a means for determining if said data communications apparatus has current access to the communications line for enabling digital data transmission during a remainder of said silent time slot scheduled for use for derived POTS signals..

14. The apparatus of claim 11, further comprising:

a means for activating a time slot for one of said analog devices when needed for service, so that transmission time can be maximized for said signal from said digital devices.

15. The apparatus of claim 11, further comprising:

a means for relocating a transmission analog time slot to be adjacent to a previous analog time slot to greatly increases bandwidth efficiency by first removing a gap time slot, and allowing said digital device time slot to be a larger contiguous data time slot

FIG. 1

(PRIOR ART)

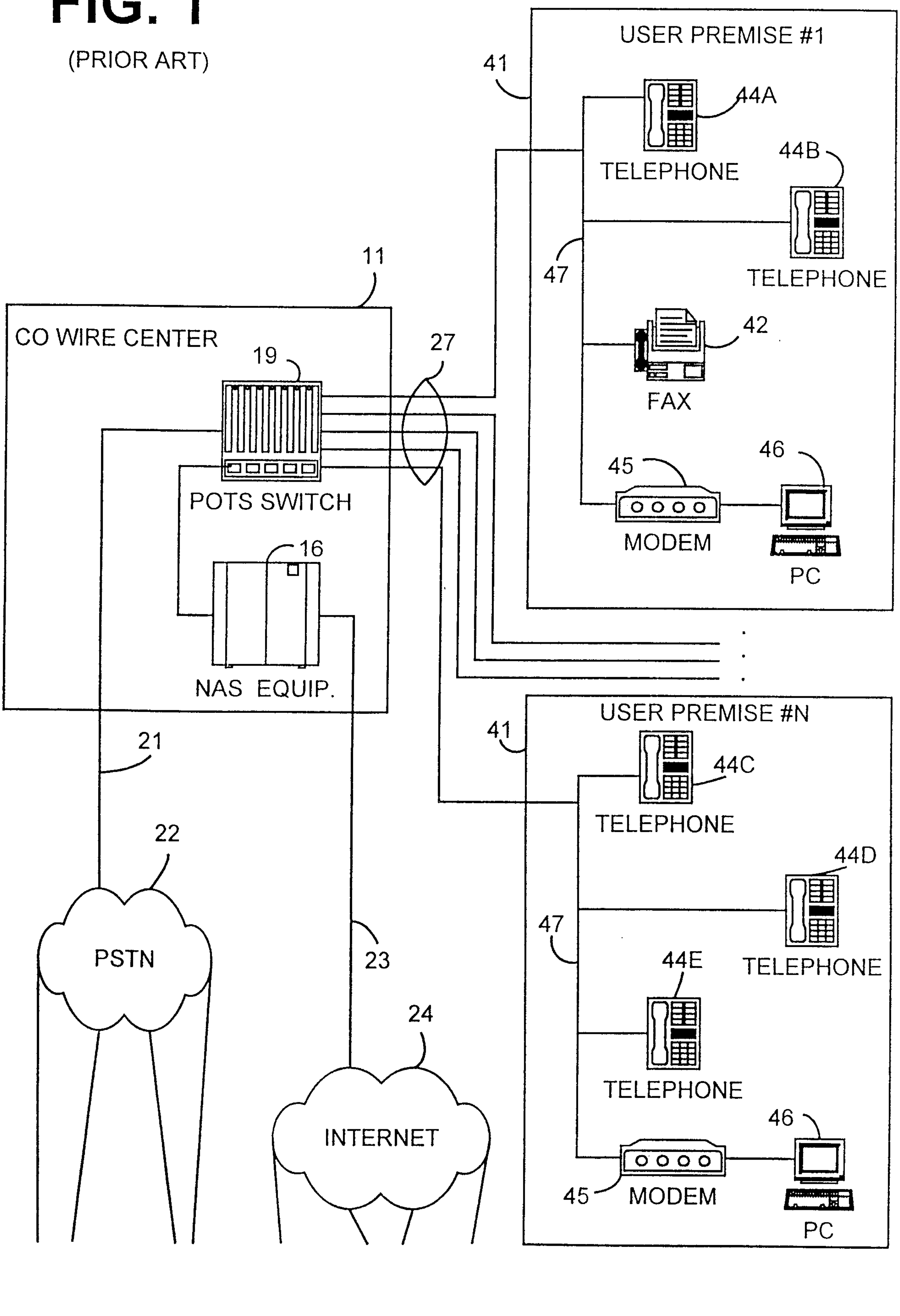


FIG. 2

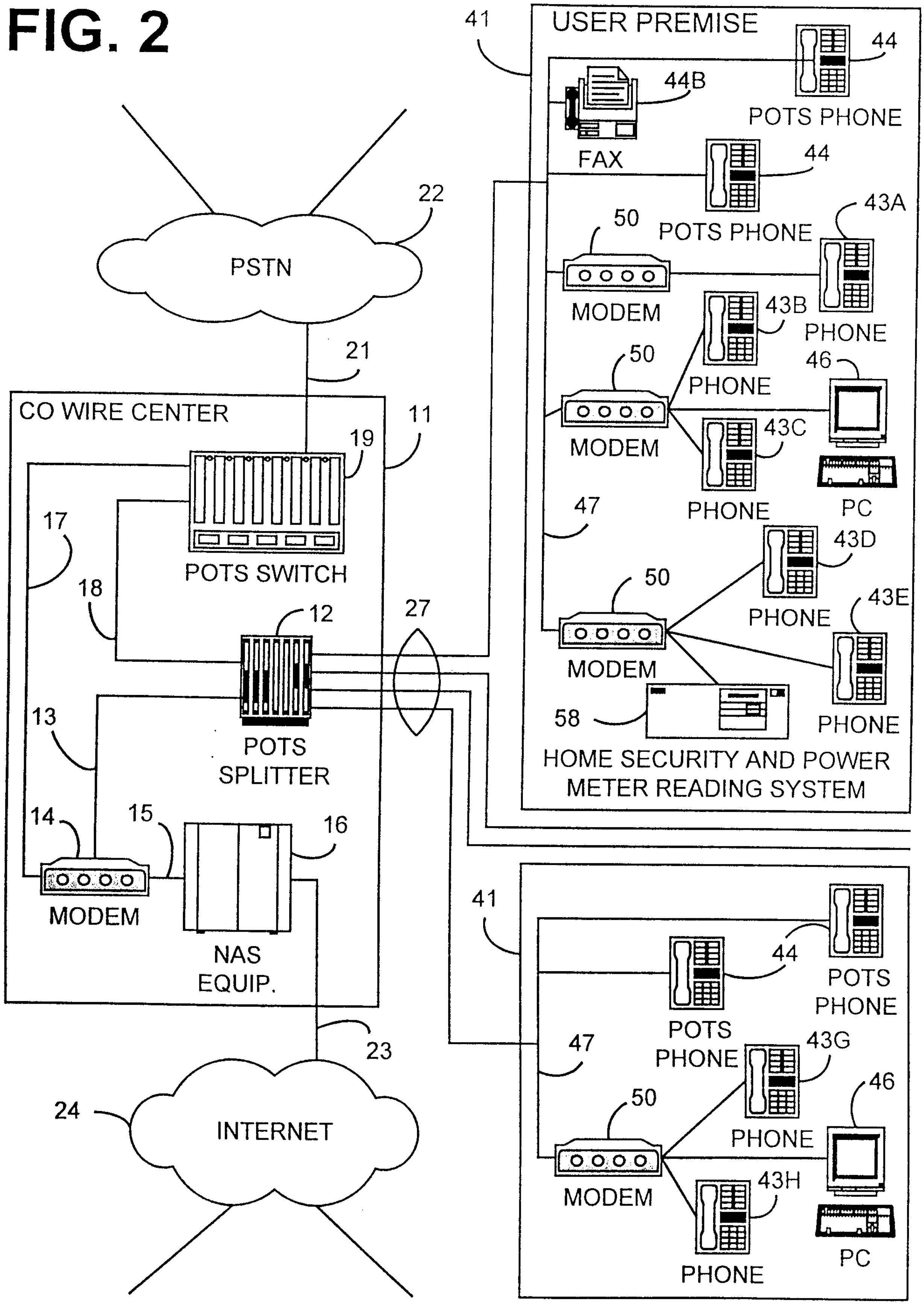
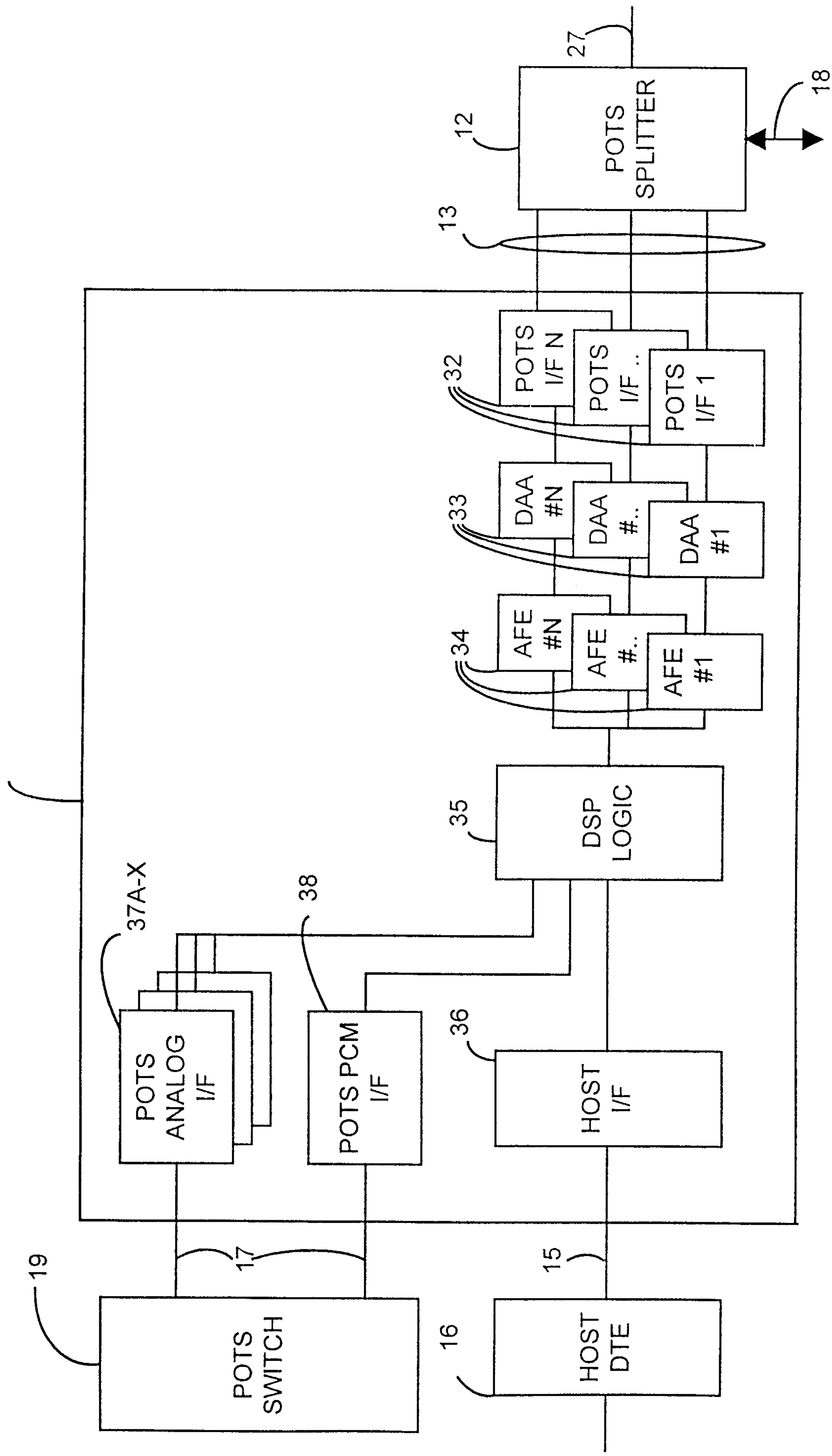

USER PREMISE
41
44
44B
POTS PHONE
FAX
44
POTS PHONE
22
50
43A
PSTN
MODEM
43B
PHONE
21
50
46
PHONE
CO WIRE CENTER
11
19
MODEM
43C
PHONE
PC
17
47
POTS SWITCH
43D
18
50
12
27
PHONE
43E
MODEM
13
58
PHONE
POTS
SPLITTER
HOME SECURITY AND POWER
METER READING SYSTEM
14
15
16
MODEM
41
44
POTS
PHONE
NAS
EQUIP.
POTS
PHONE
23
47
43G
50
46
24
INTERNET
PHONE
MODEM
43H
PC
PHONE

FIG. 3A



27
12
POTS SPLITTER
18
13
14
POTS I/F N
POTS I/F ..
POTS I/F 1
32
DAA #N
DAA #..
DAA #1
33
AFE #N
AFE #..
AFE #1
34
35
DSP LOGIC
37A-X
38
POTS ANALOG I/F
POTS PCM I/F
36
HOST I/F
17
15
19
POTS SWITCH
16
HOST DTE

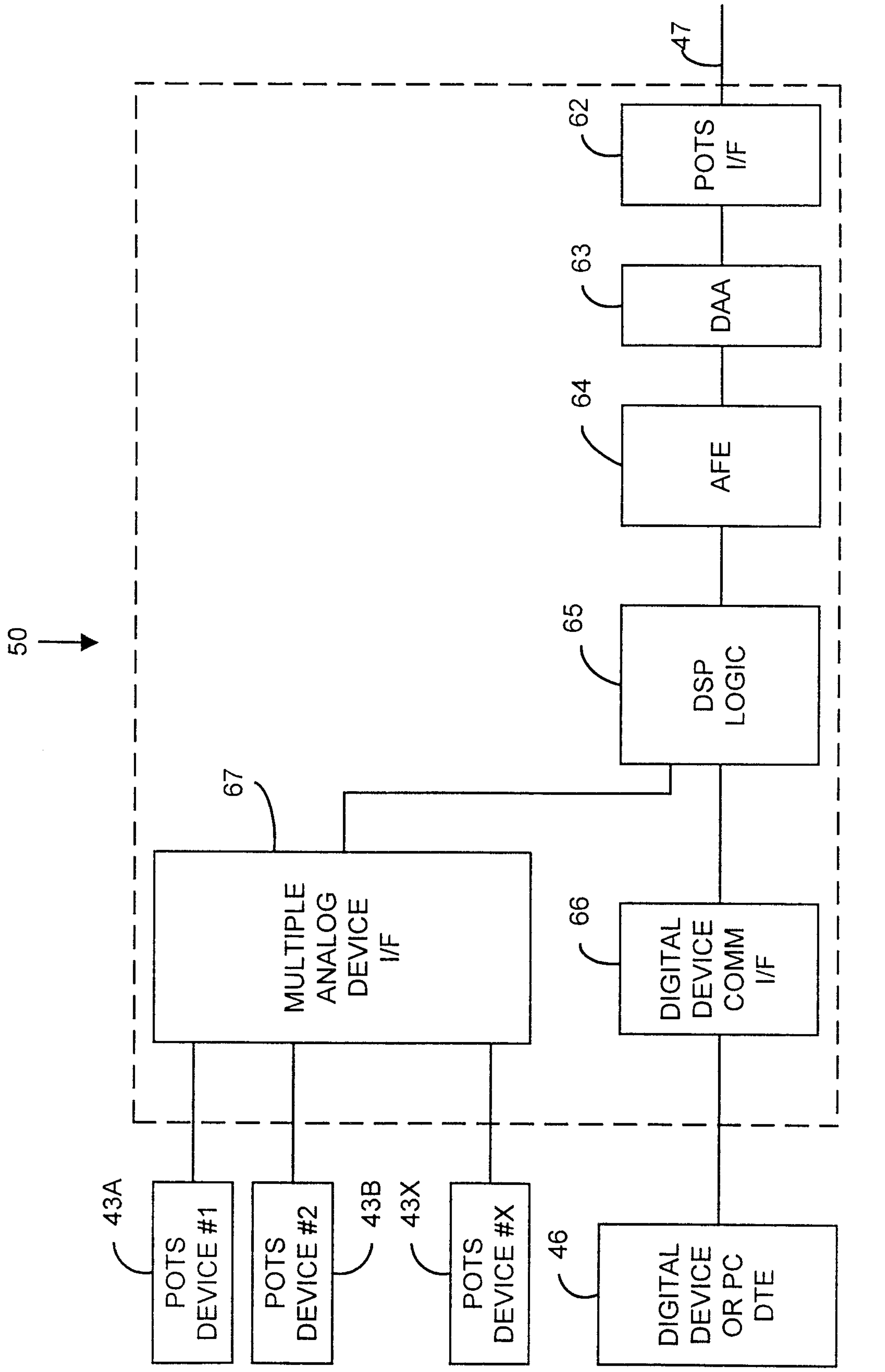

50
47
62
POTS I/F
63
DAA
64
AFE
65
DSP LOGIC
67
MULTIPLE ANALOG DEVICE I/F
66
DIGITAL DEVICE COMM I/F
43A
POTS DEVICE #1
43B
POTS DEVICE #2
43X
POTS DEVICE #X
46
DIGITAL DEVICE OR PC DTE

FIG. 3B

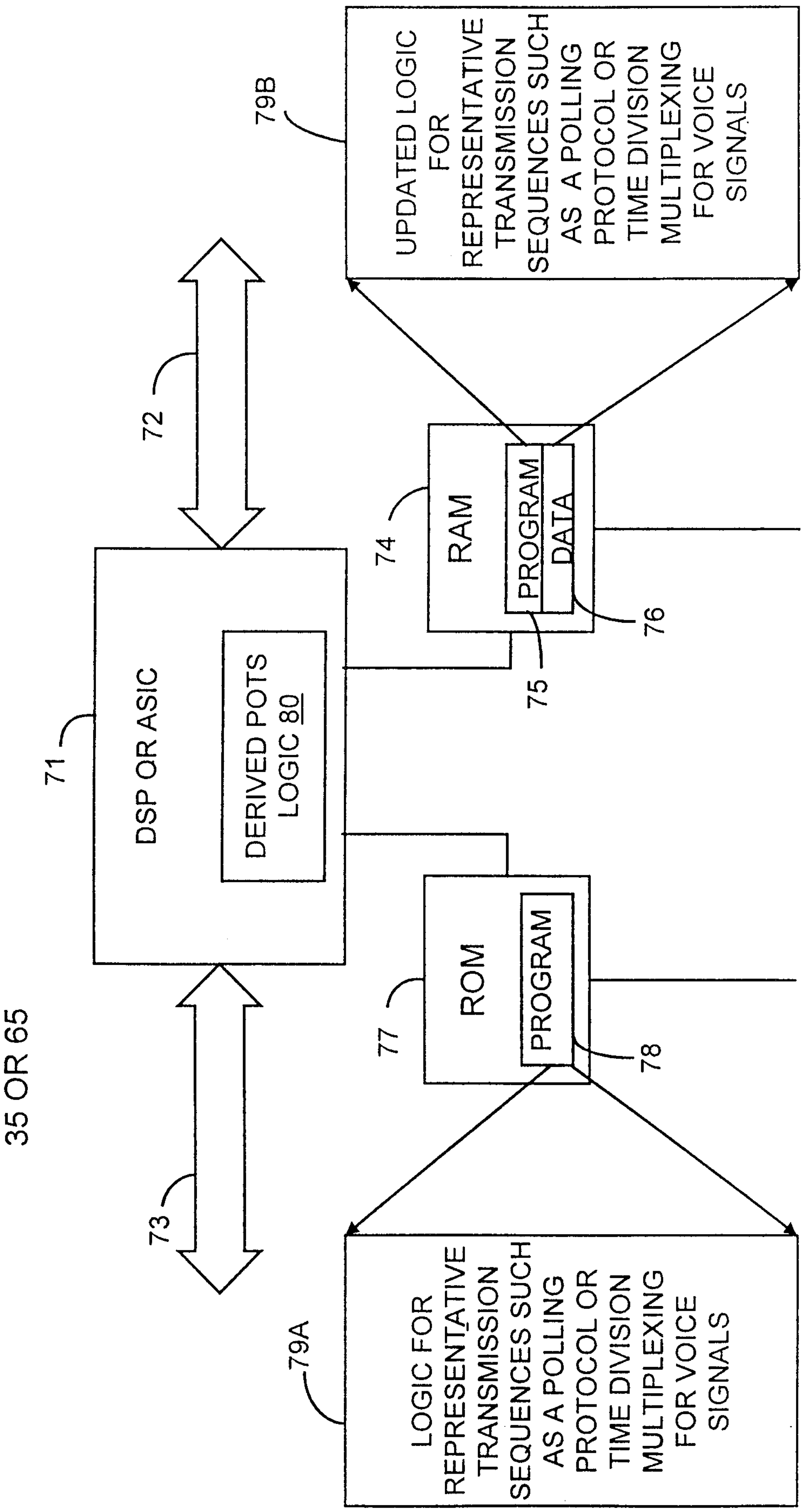

35 OR 65
71
DSP OR ASIC
DERIVED POTS LOGIC 80
72
73
74
RAM
PROGRAM
DATA
75
76
77
ROM
PROGRAM
78
79A
LOGIC FOR REPRESENTATIVE TRANSMISSION SEQUENCES SUCH AS A POLLING PROTOCOL OR TIME DIVISION MULTIPLEXING FOR VOICE SIGNALS
79B
UPDATED LOGIC FOR REPRESENTATIVE TRANSMISSION SEQUENCES SUCH AS A POLLING PROTOCOL OR TIME DIVISION MULTIPLEXING FOR VOICE SIGNALS

FIG. 4

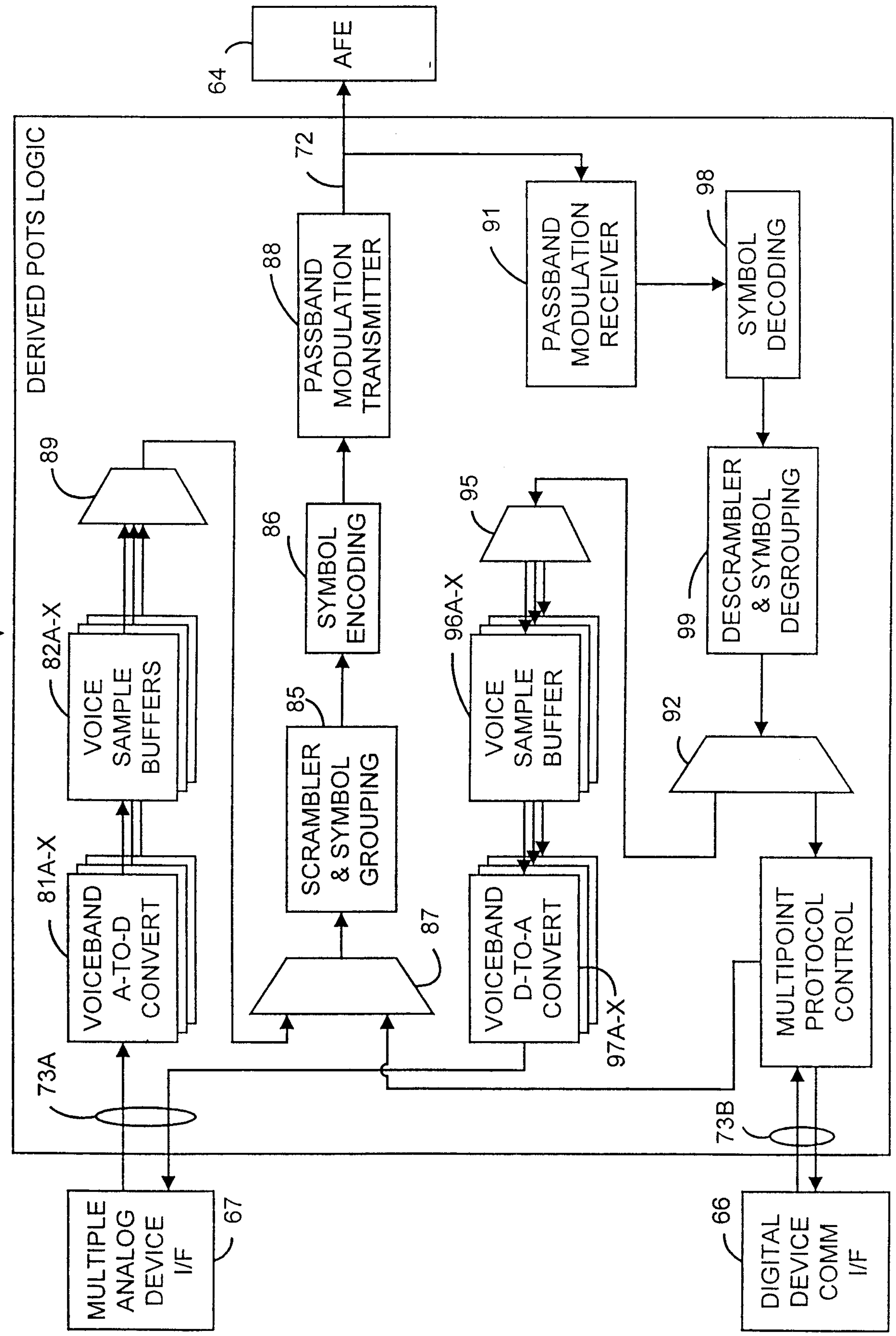

AFE
64
DERIVED POTS LOGIC
80
72
PASSBAND MODULATION TRANSMITTER
88
PASSBAND MODULATION RECEIVER
91
SYMBOL DECODING
98
89
SYMBOL ENCODING
86
95
DESCRAMBLER & SYMBOL DEGROUPING
99
82A-X
VOICE SAMPLE BUFFERS
96A-X
VOICE SAMPLE BUFFER
85
SCRAMBLER & SYMBOL GROUPING
92
81A-X
VOICEBAND A-TO-D CONVERT
87
VOICEBAND D-TO-A CONVERT
97A-X
MULTIPOINT PROTOCOL CONTROL
73A
73B
MULTIPLE ANALOG DEVICE I/F
67
DIGITAL DEVICE COMM I/F
66

FIG. 5

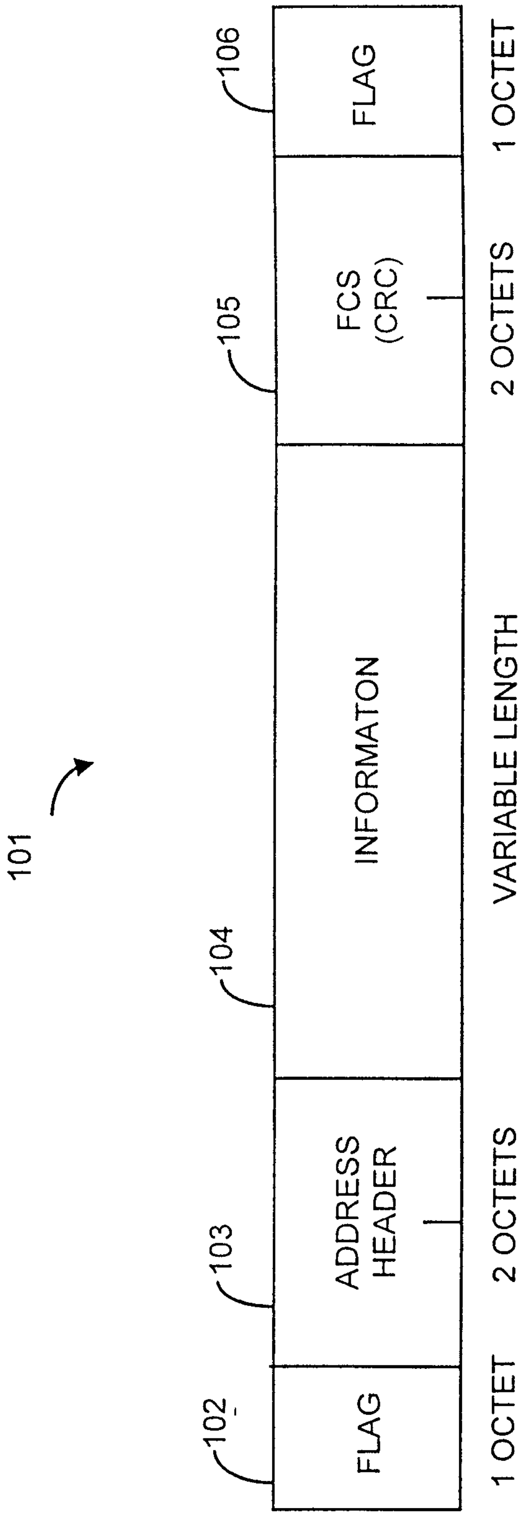

101
102
FLAG
1 OCTET
103
ADDRESS
HEADER
2 OCTETS
104
INFORMATON
VARIABLE LENGTH
105
FCS
(CRC)
2 OCTETS
106
FLAG
1 OCTET

Fig. 6

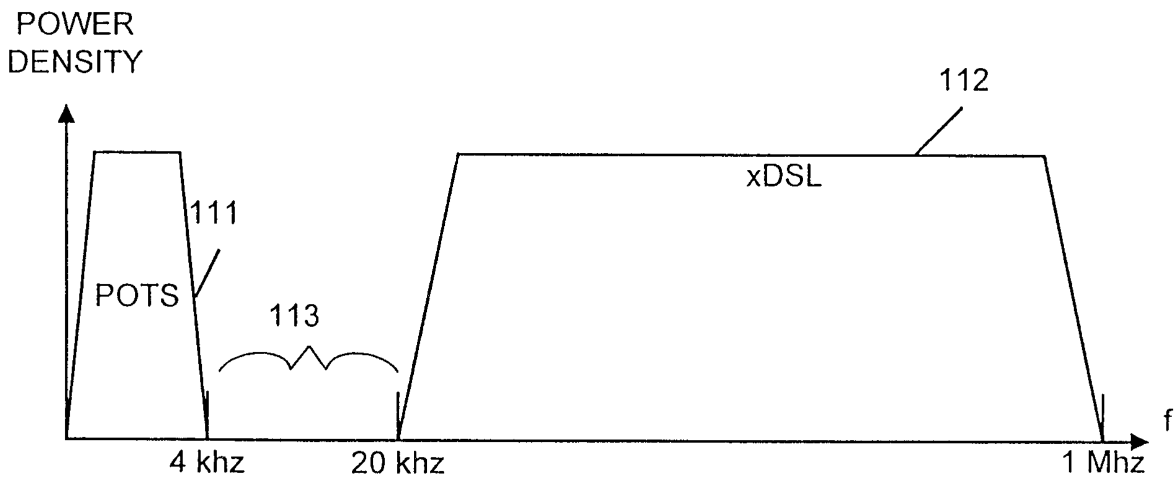

POWER
DENSITY
112
xDSL
111
POTS
113
f
4 khz
20 khz
1 Mhz

FIG. 7

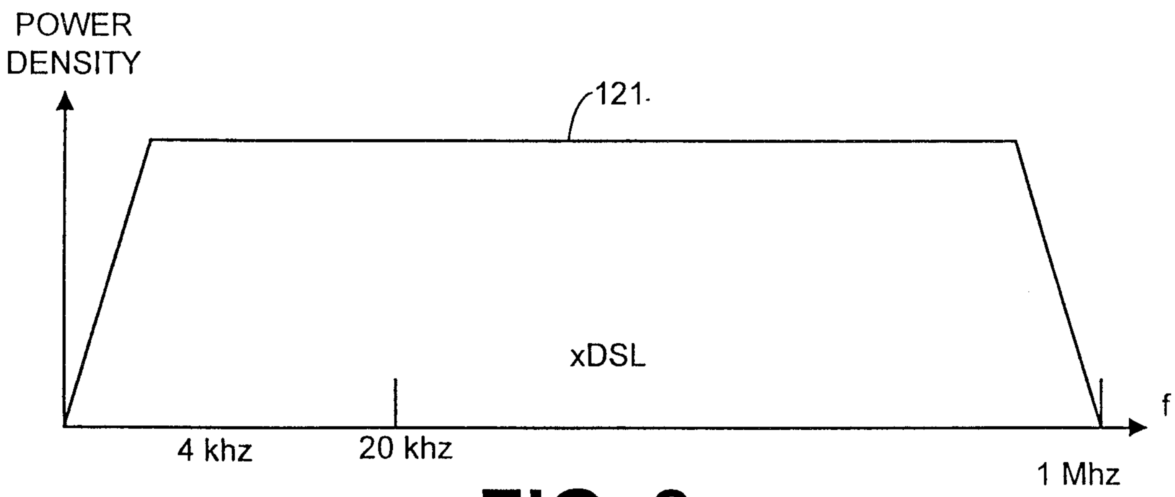

POWER
DENSITY
121
xDSL
f
4 khz
20 khz
1 Mhz

FIG. 8

Fig. 9A

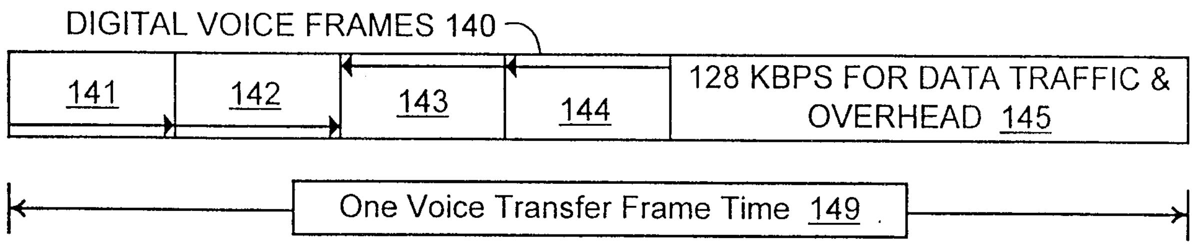


Fig. 9B

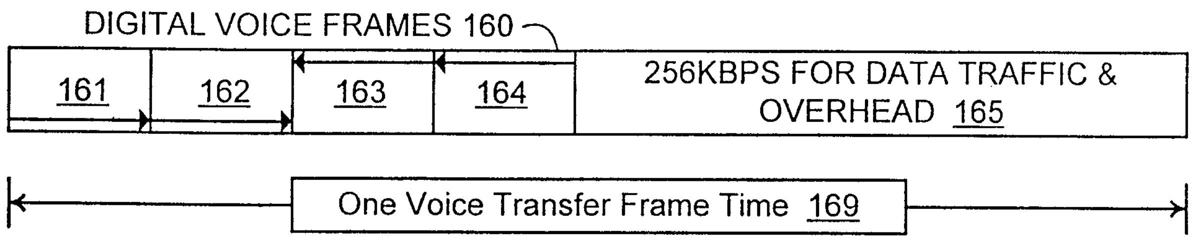


Fig. 9C

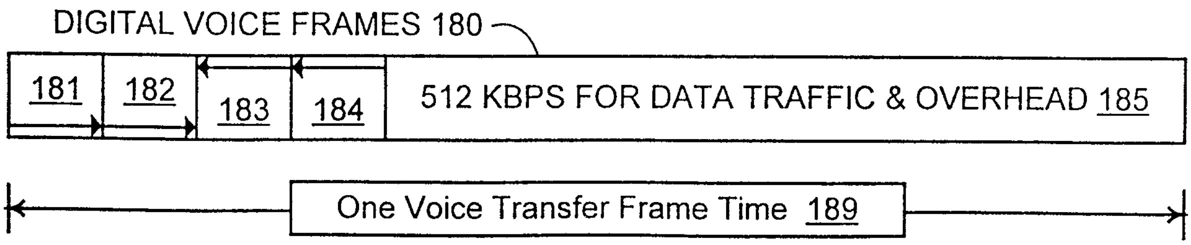


Fig. 10A

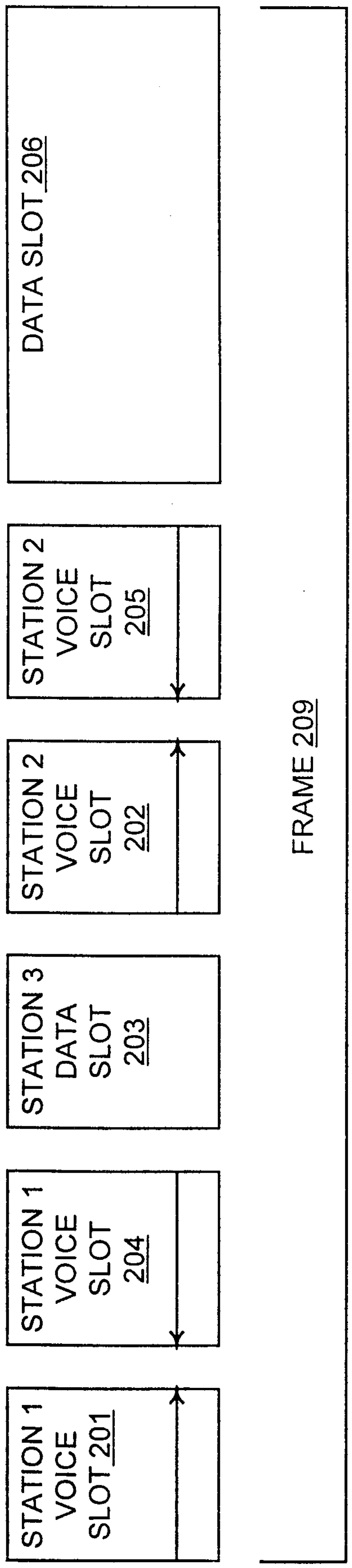
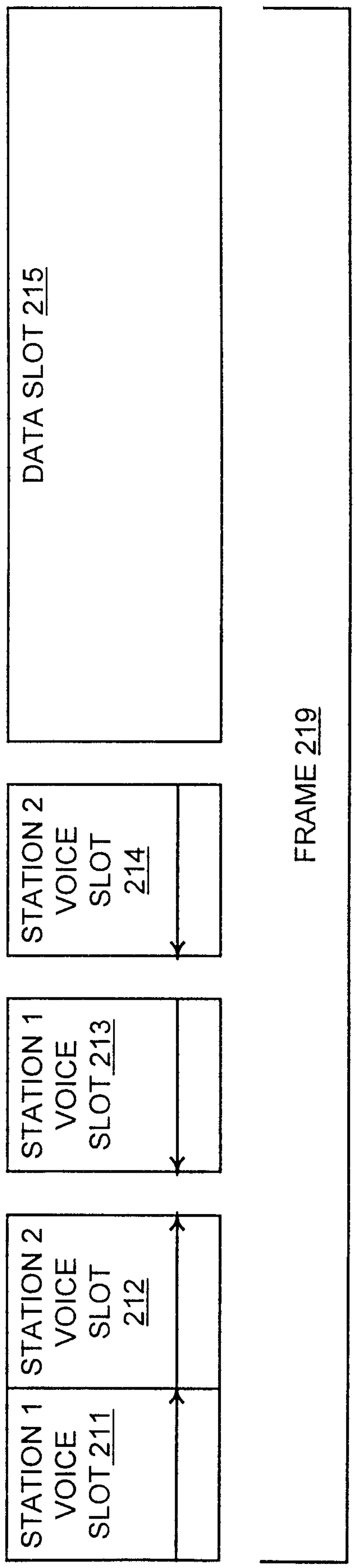


Fig. 10B



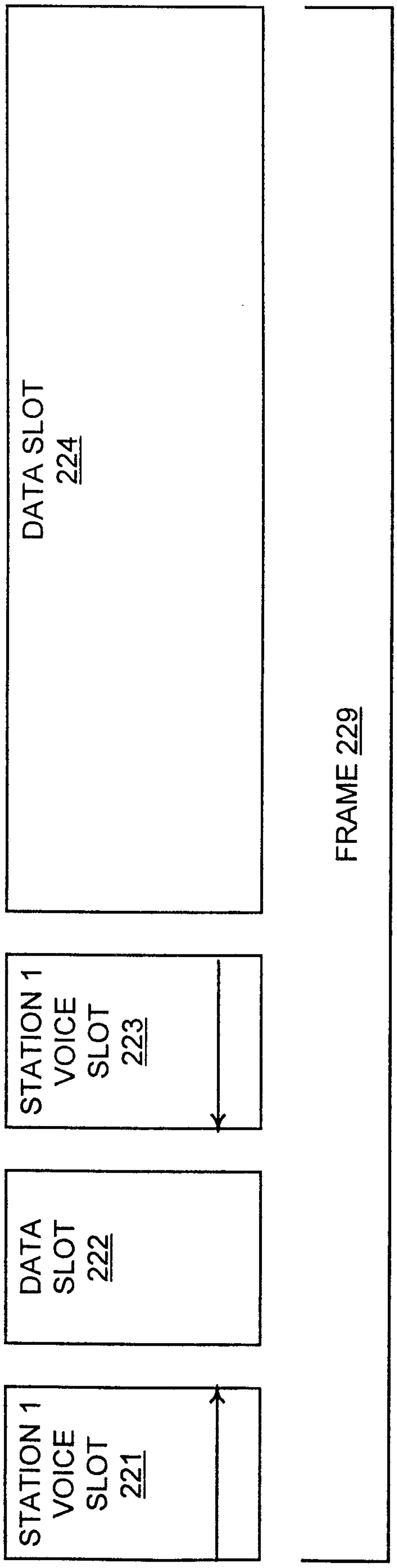


Fig. 10C

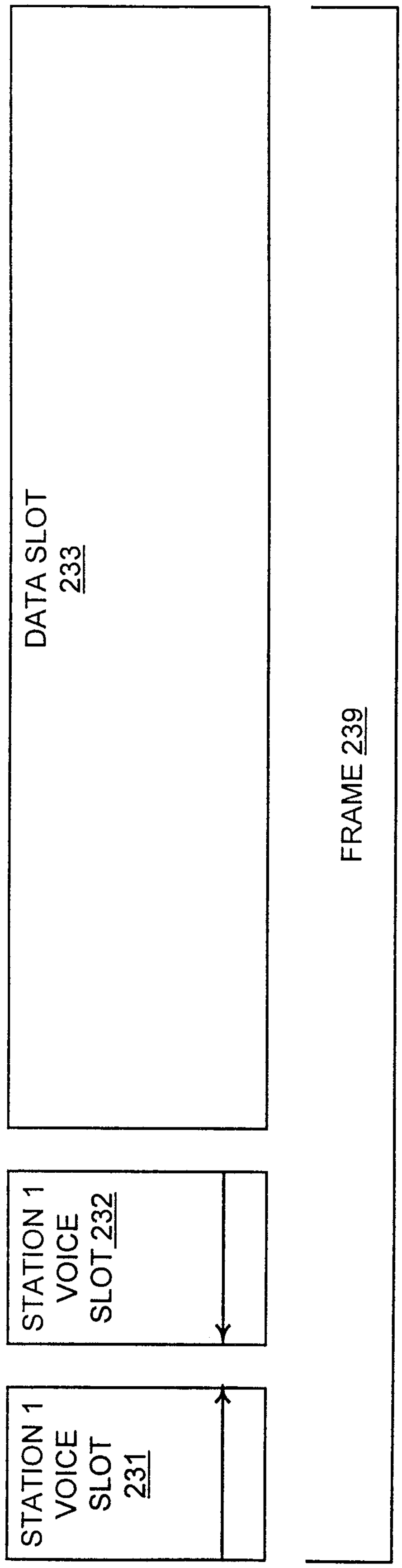
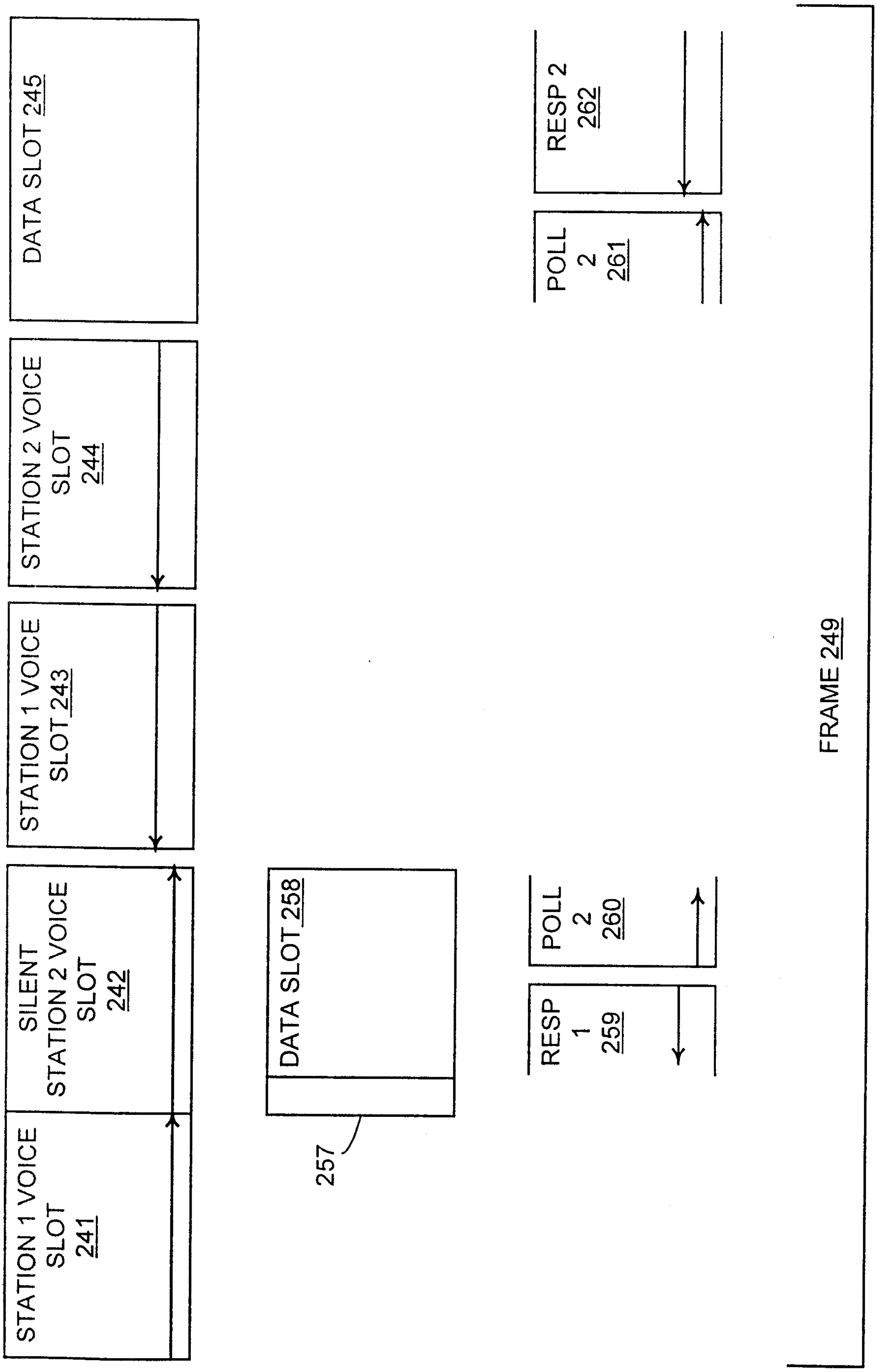


Fig. 10D

Fig. 10E


STATION 1 VOICE SLOT 241
SILENT STATION 2 VOICE SLOT 242
STATION 1 VOICE SLOT 243
STATION 2 VOICE SLOT 244
DATA SLOT 245
257
DATA SLOT 258
RESP 1 259
POLL 2 260
POLL 2 261
RESP 2 262
FRAME 249