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IN THE UNITED STATES DISTRICT COURT
MIDDLE DISTRICT OF FLORIDA
TAMPA DIVISION

PARADYNE CORPORATION, a Delaware Corporation,	)	
Plaintiff,	)	Civil Action File No.: 8:04CV1278-T24 TGW
v.	)	
GENERAL BANDWIDTH, INC. a Delaware Corporation,	)	
Defendant.	)	

COMPLAINT AND DEMAND FOR JURY TRIAL

The Plaintiff, Paradyne Corporation, states as its Complaint as follows:

JURISDICTION AND VENUE

1. This is an action for patent infringement arising under the patent laws of the United States, Title 35, United States Code.

2. This Court has subject matter jurisdiction over all causes of action set forth herein pursuant to 28 U.S.C. §§1331 and 1338(a).

3. Venue is proper in this judicial district and division pursuant to 28 U.S.C. §§1391(b) and (c).

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#150

SCANNED

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THE PARTIES

4. Plaintiff, Paradyne Corporation ("Paradyne"), is a Delaware Corporation, with a principal place of business at 8545 126th Avenue North, Largo, Florida 33773.

5. Paradyne is the owner, by assignment, of all right, title, and interest in and to United States Patent No. 6,639,913, (Exhibit "A"), United States Patent No. 6,580,785 (Exhibit "B"), United States Patent No. 6,546,090 (Exhibit "C"), United States Patent No. 6,154,524 (Exhibit "D"), United States Patent No. 6,111,936 (Exhibit "E"), United States Patent No. 6,075,784 (Exhibit "F"), United States Patent No. 6,061,392 (Exhibit "G"), and United States Patent No. 5,826,034 (Exhibit "H")(all eight patents are hereinafter referred to as "the Paradyne patents-in-suit").

6. Upon information and belief, Defendant, General Bandwidth, Inc. ("Defendant") is a Delaware Corporation, with a principal place of business at 12303 Technology Boulevard, Austin, Texas 78727-6104.

7. Upon information and belief, Defendant has and continues to infringe the patents-in-suit in the State of Florida, within this judicial district, and elsewhere throughout the United States. Upon information and belief, Defendant is subject to the personal jurisdiction of this Court.

THE CONTROVERSY

8. Defendant has in the past and continues to make, have made, offer for sale, sell, and import into the United States one or more products and/or processes that infringe one or more claims of each of the patents-in-suit. For example, Defendant's "G6" gateway has and continues to infringe one or more claims of each of the Paradyne patents-in-suit.

9. Paradyne placed Defendant on notice of its on-going infringement of U.S. Patent Nos. 6,580,785, 6,546,090, 6,075,784, and 6,061,392 on September 10, 2003.

COUNT ONE: PATENT INFRINGEMENT

10. Paradyne realleges and incorporates herein the allegations of paragraphs 1 through 9 of this Complaint as if fully set forth herein.

11. Upon information and belief, Defendant has engaged in the manufacture, use, offering for sale, sale, and/or importation of products and/or processes that constitute direct infringement, contributory infringement, and/or inducement to infringe one or more claims of the Paradyne patents-in-suit in violation of 35 U.S.C. §271. The Defendant's infringing products and/or processes manufactured, used, offered for sale, sold, and/or imported into the

United States include, but are not necessarily limited to, Defendant's "G6" gateway product.

12. Defendant was notified of its on-going infringement of U.S. Patent Nos. 6,580,785, 6,546,090, 6,075,784, and 6,061,392 on September 10, 2003. See Exhibit "I."

13. Defendant's infringement of the Paradyne patents-in-suit has been, and continues to be, willful.

14. Paradyne has and continues to suffer damages as a direct and proximate result of Defendant's infringement of the Paradyne patents-in-suit, and will suffer additional and irreparable damages unless Defendant is permanently enjoined by this Court from continuing its infringement. Paradyne has no adequate remedy at law.

15. Paradyne is entitled to: (i) damages adequate to compensate it for Defendant's infringement, which amounts to, at a minimum, a reasonable royalty; (ii) treble damages; (iii) its attorneys' fees and costs; and (iv) a preliminary and permanent injunction.

PRAYER FOR RELIEF

WHEREFORE, Paradyne seeks the following relief:

a. that Defendant be ordered to pay damages adequate to compensate Paradyne for Defendant's infringement of the Paradyne patents-in-suit pursuant to 35 U.S.C. §284;

b. that Defendant be ordered to pay treble damages and attorneys' fees pursuant to 35 U.S.C. §§284 and 285;

c. that Defendant be enjoined from further infringement of the Paradyne patents-in-suit pursuant to 35 U.S.C. §283;

d. that Defendant be ordered to pay prejudgment interest;

e. that Defendant be ordered to pay all costs associated with this action; and

f. that Paradyne be granted such other and additional relief as the Court deems just and proper.

DEMAND FOR JURY TRIAL

Pursuant to Fed. R. Civ. P. 38(b), Paradyne demands a trial by jury of all issues triable of right by a jury.

THIS 4th day of June, 2004.

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DOCUMENT, ATTACHMENTS, OR EXHIBITS NOT SCANNED

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US006639913B1

(12) **United States Patent**
Frankel et al.

(10) **Patent No.: US 6,639,913 B1**
(45) **Date of Patent: *Oct. 28, 2003**

(54) **SYSTEM AND METHOD FOR COMMUNICATING VOICE AND DATA OVER A LOCAL PACKET NETWORK**

(75) Inventors: **David P. Frankel**, Sunnyvale, CA (US); **Joe Boucher**, Sunnyvale, CA (US); **Kenneth M. Kolderup**, Saratoga, CA (US)

(73) Assignee: **Paradyne Corporation**, Largo, FL (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

This patent is subject to a terminal disclaimer.

(21) Appl. No.: **09/314,318**

(22) Filed: **May 19, 1999**

Related U.S. Application Data

(63) Continuation-in-part of application No. 09/112,911, filed on Jul 9, 1998, now Pat. No. 6,075,784.

(60) Provisional application No. 60/088,399, filed on Jun. 8, 1998.

(51) **Int. Cl.**[7] **H04L 12/66**

(52) **U.S. Cl.** **370/356**; 370/352

(58) **Field of Search** 370/352, 353, 370/354, 355, 356, 466, 467, 357, 386–388; 455/3.1–3.3, 4.1–4.2, 5.1–6.2; 379/88.17

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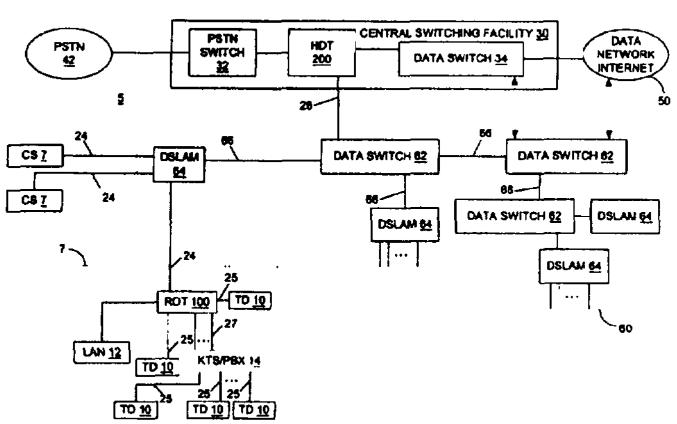
(List continued on next page.)

Primary Examiner—Dang Ton
Assistant Examiner—Phuc Tran
(74) *Attorney, Agent, or Firm*—Thomas, Kayden, Horstemeyer & Risley, L.L.P.

(57) **ABSTRACT**

A system and method of communicating voice and data via a local packet network (LPN) to and from a customer site. A remote digital terminal (RDT) is provided at a customer site to interface a plurality of telephone devices and/or data devices (computers or a local area network of computers) with the LPN via a local loop link, such as a Digital Subscriber Line or a wireless local loop. A host digital terminal (HDT) is provided at a control site within or connected to the LPN that coordinates the communication of voice calls between the RDT and a public switched telephone network (PSTN) switch via the LPN and that coordinates the communication of data between the RDT and a data network within or without the LPN. Multiple telephone calls with the customer site can be supported by the remote digital terminal over a single local loop link connected to the LPN. A wire center remote digital terminal (WC-RDT) connects to or is incorporated in a digital subscriber line access multiplexer (DSLAM) that is connected to or part of the LPN. The WC-RDT provides standard analog telephone service between a plurality of standard telephone ports suitable for connection to local loop links of a plurality of customer sites and the PSTN switch through the host digital terminal. The WC-RDT is used to provide POTS service to customer sites having an RDT as well as to customer sites that do not have an RDT.

38 Claims, 8 Drawing Sheets



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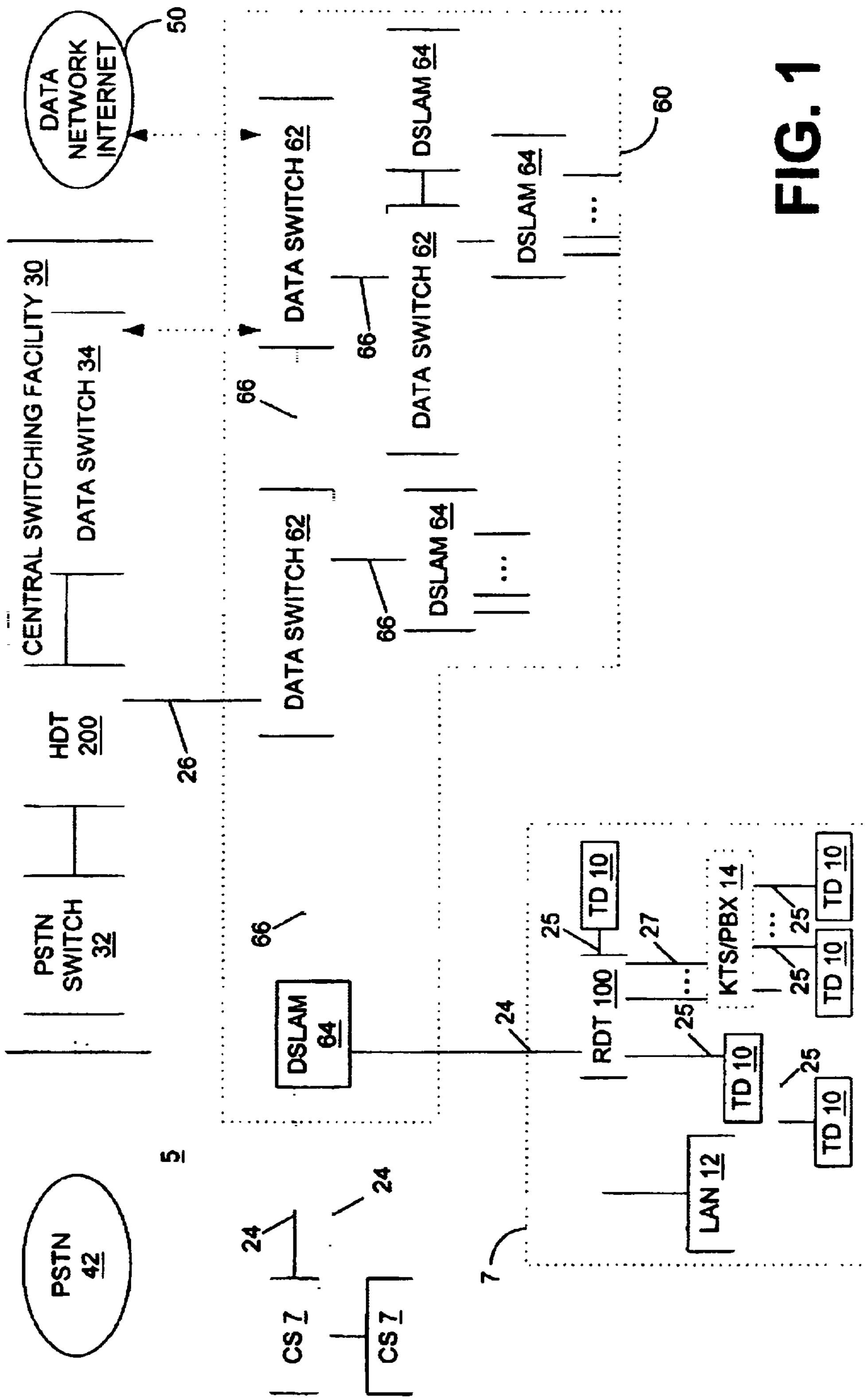

FIG. 1

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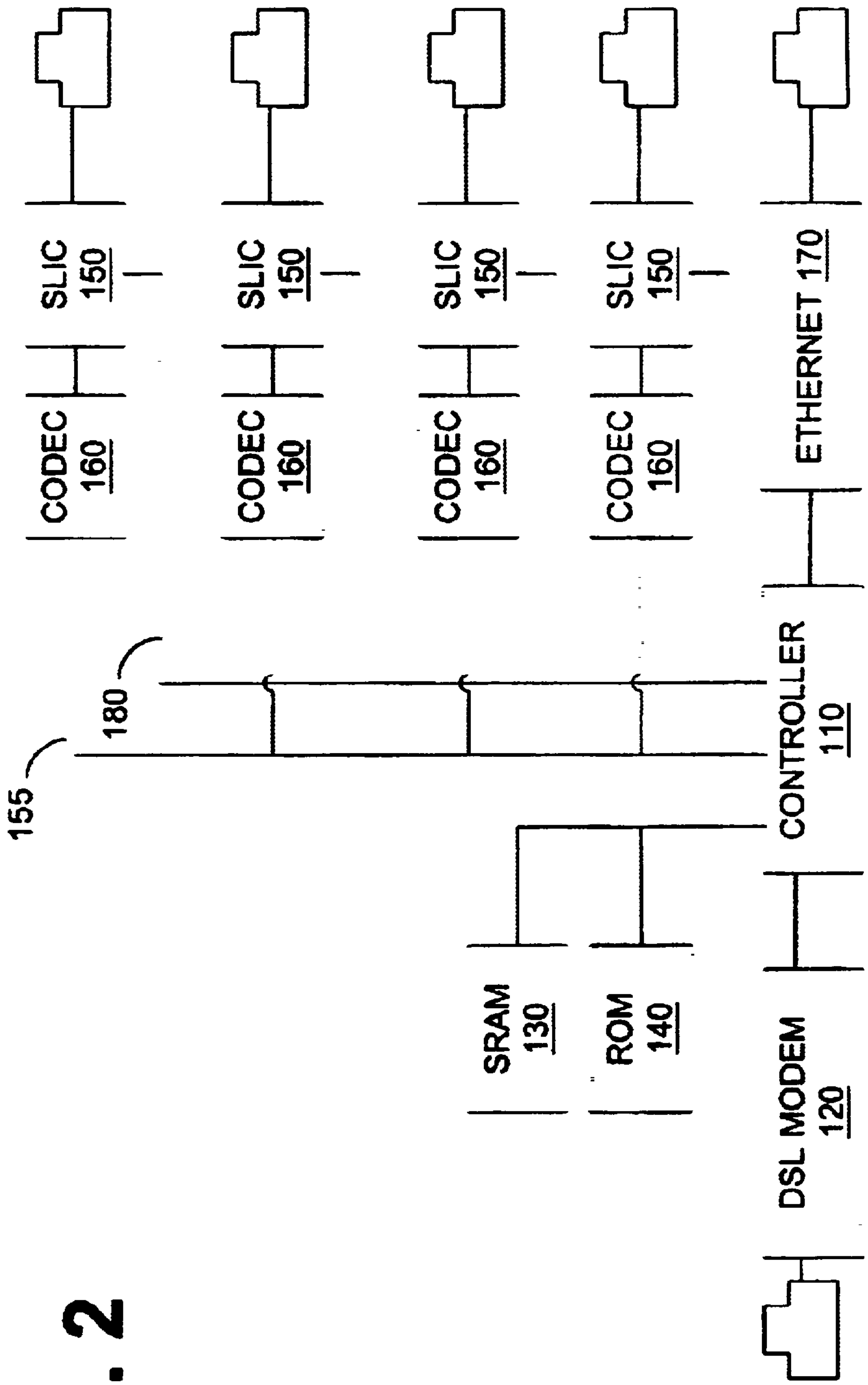

FIG. 2

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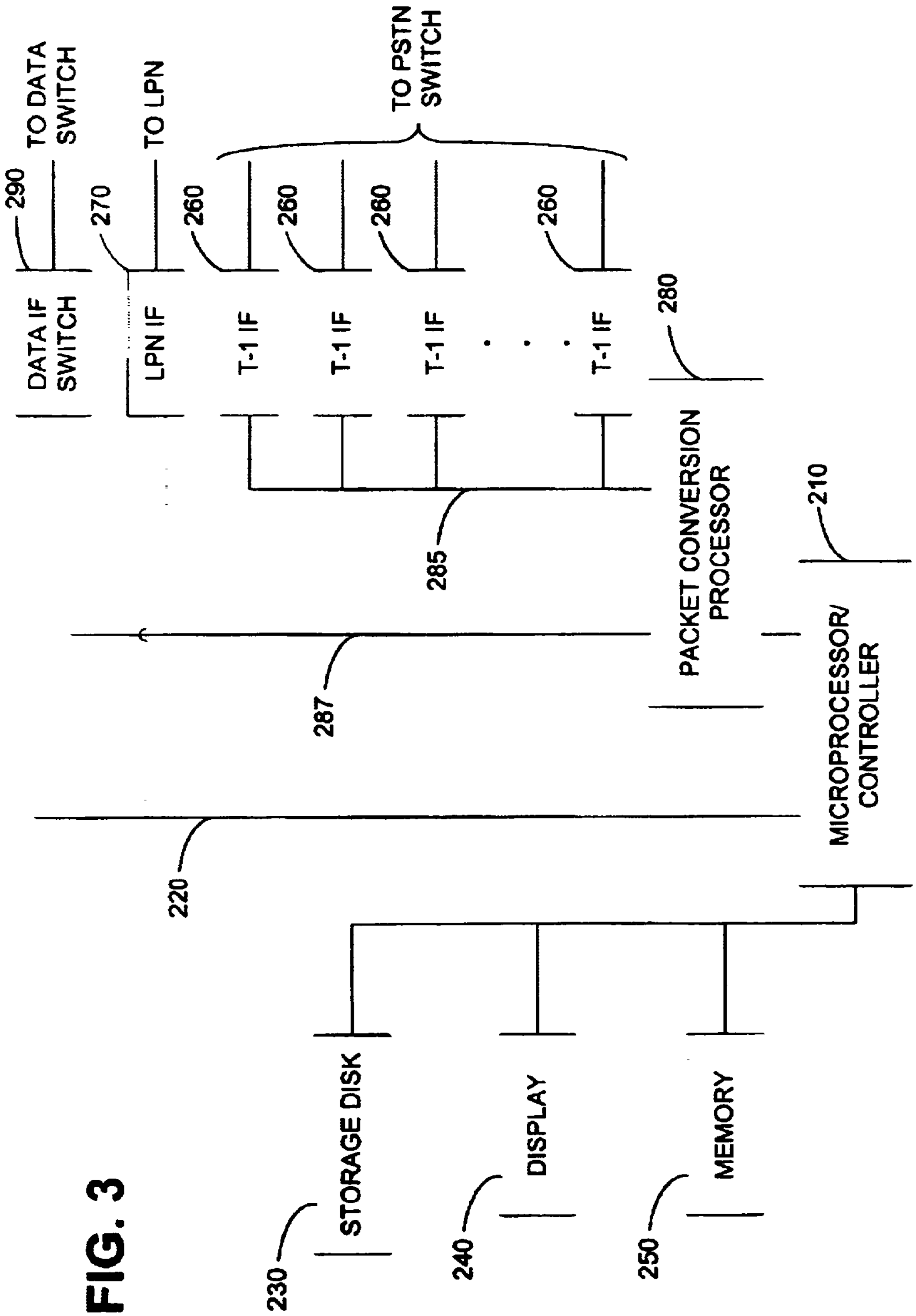
TO DATA SWITCH
TO LPN
TO PSTN SWITCH
290
270
260
260
260
260
DATA IF SWITCH
LPN IF
T-1 IF
T-1 IF
T-1 IF
T-1 IF
280
PACKET CONVERSION PROCESSOR
285
210
287
MICROPROCESSOR/ CONTROLLER
220
STORAGE DISK
DISPLAY
MEMORY
230
240
250

FIG. 3

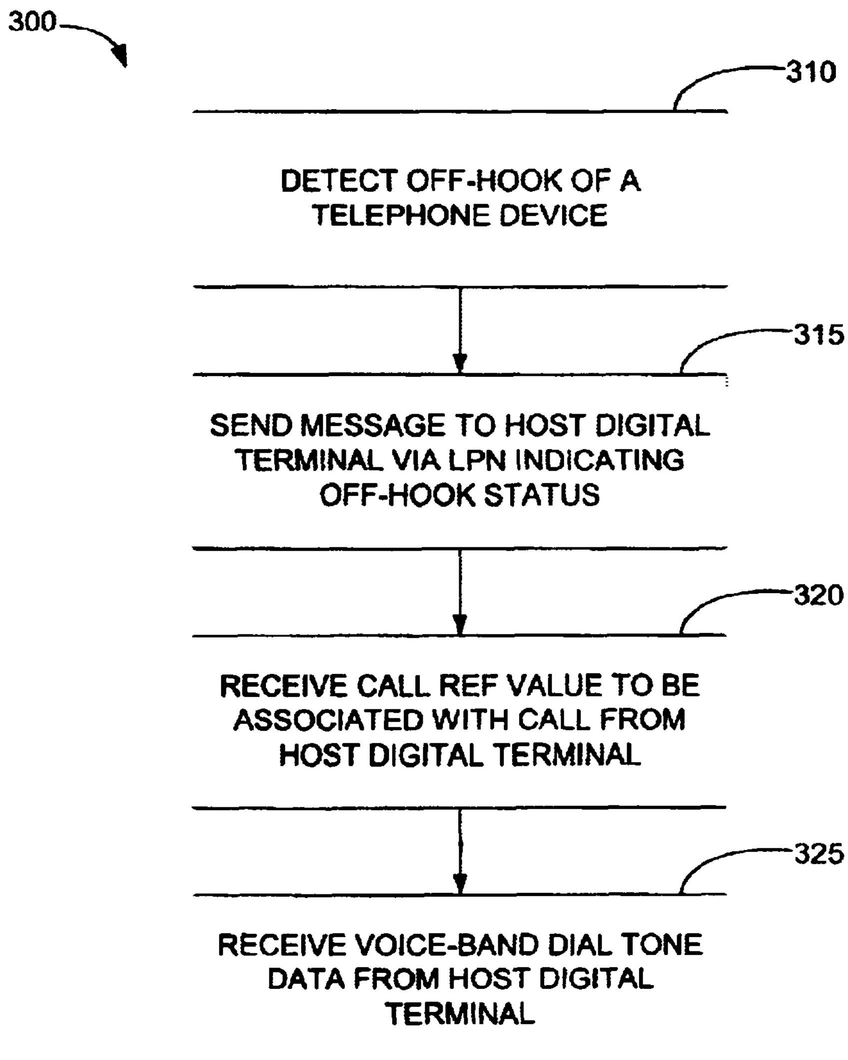

FIG. 4

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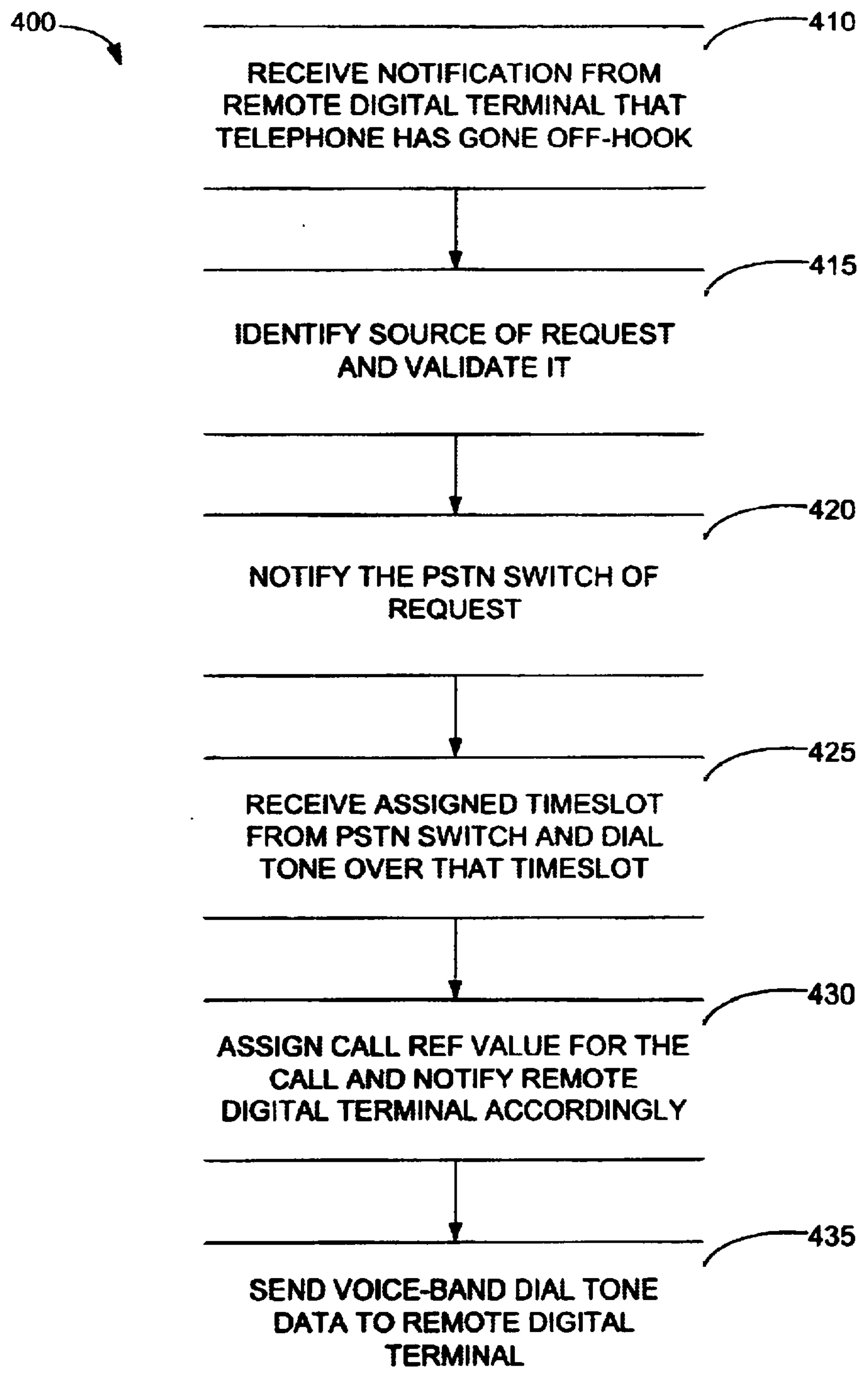
400
410
RECEIVE NOTIFICATION FROM REMOTE DIGITAL TERMINAL THAT TELEPHONE HAS GONE OFF-HOOK
415
IDENTIFY SOURCE OF REQUEST AND VALIDATE IT
420
NOTIFY THE PSTN SWITCH OF REQUEST
425
RECEIVE ASSIGNED TIMESLOT FROM PSTN SWITCH AND DIAL TONE OVER THAT TIMESLOT
430
ASSIGN CALL REF VALUE FOR THE CALL AND NOTIFY REMOTE DIGITAL TERMINAL ACCORDINGLY
435
SEND VOICE-BAND DIAL TONE DATA TO REMOTE DIGITAL TERMINAL

FIG. 5

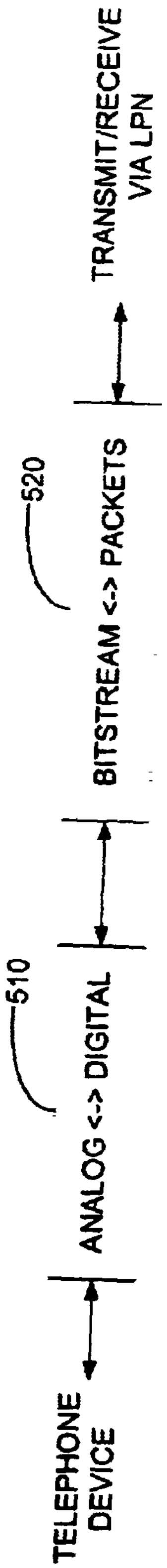
TELEPHONE DEVICE
ANALOG <-> DIGITAL
510
BITSTREAM <-> PACKETS
520
TRANSMIT/RECEIVE VIA LPN

FIG. 6

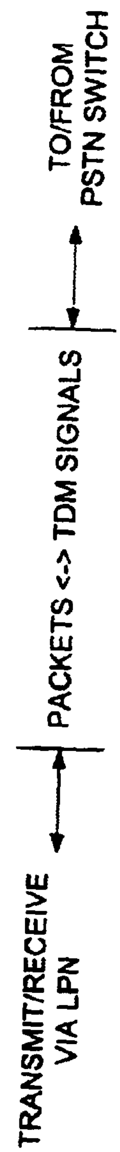
TRANSMIT/RECEIVE VIA LPN
PACKETS <-> TDM SIGNALS
TO/FROM PSTN SWITCH

FIG. 7

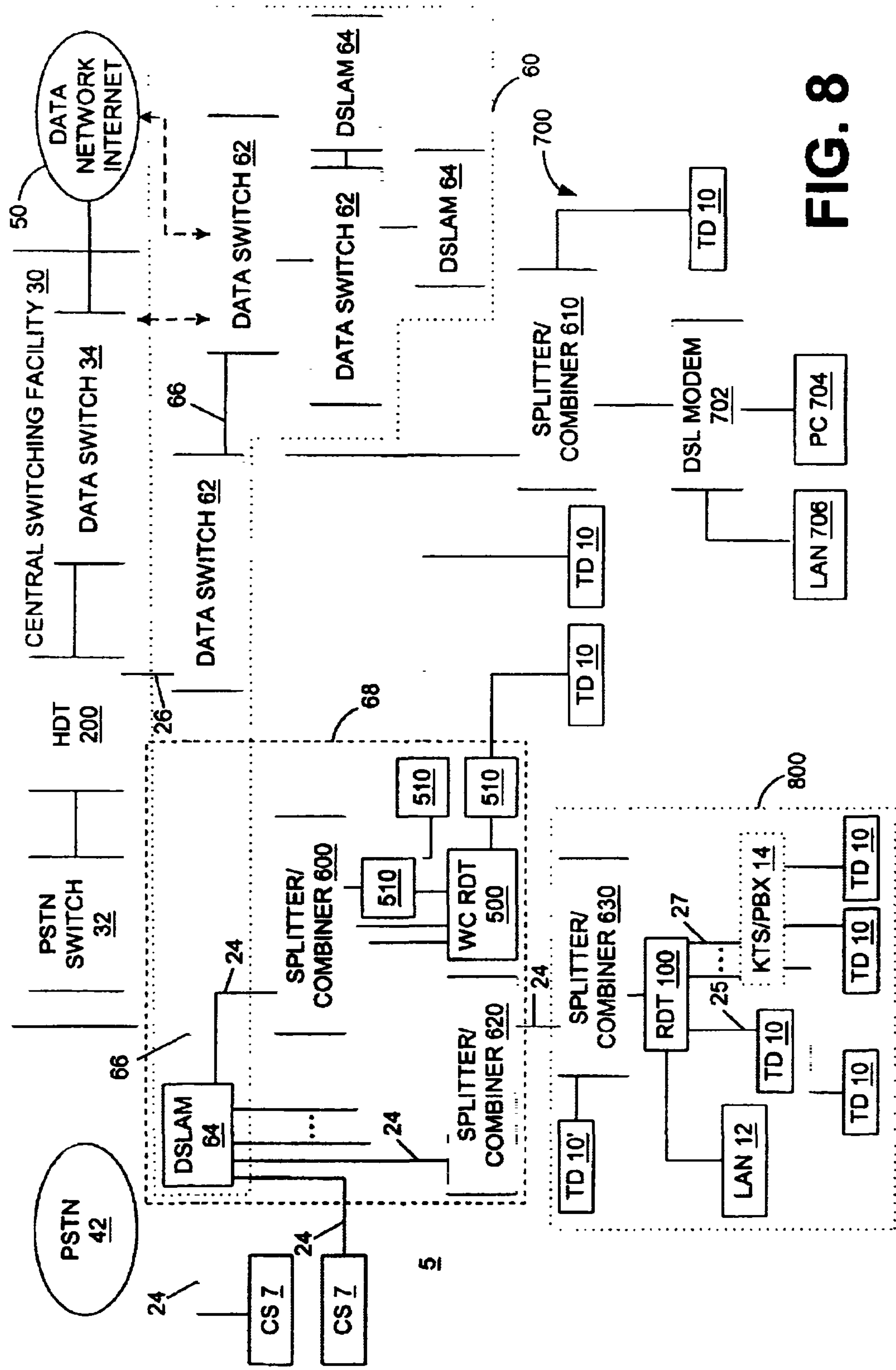

FIG. 8

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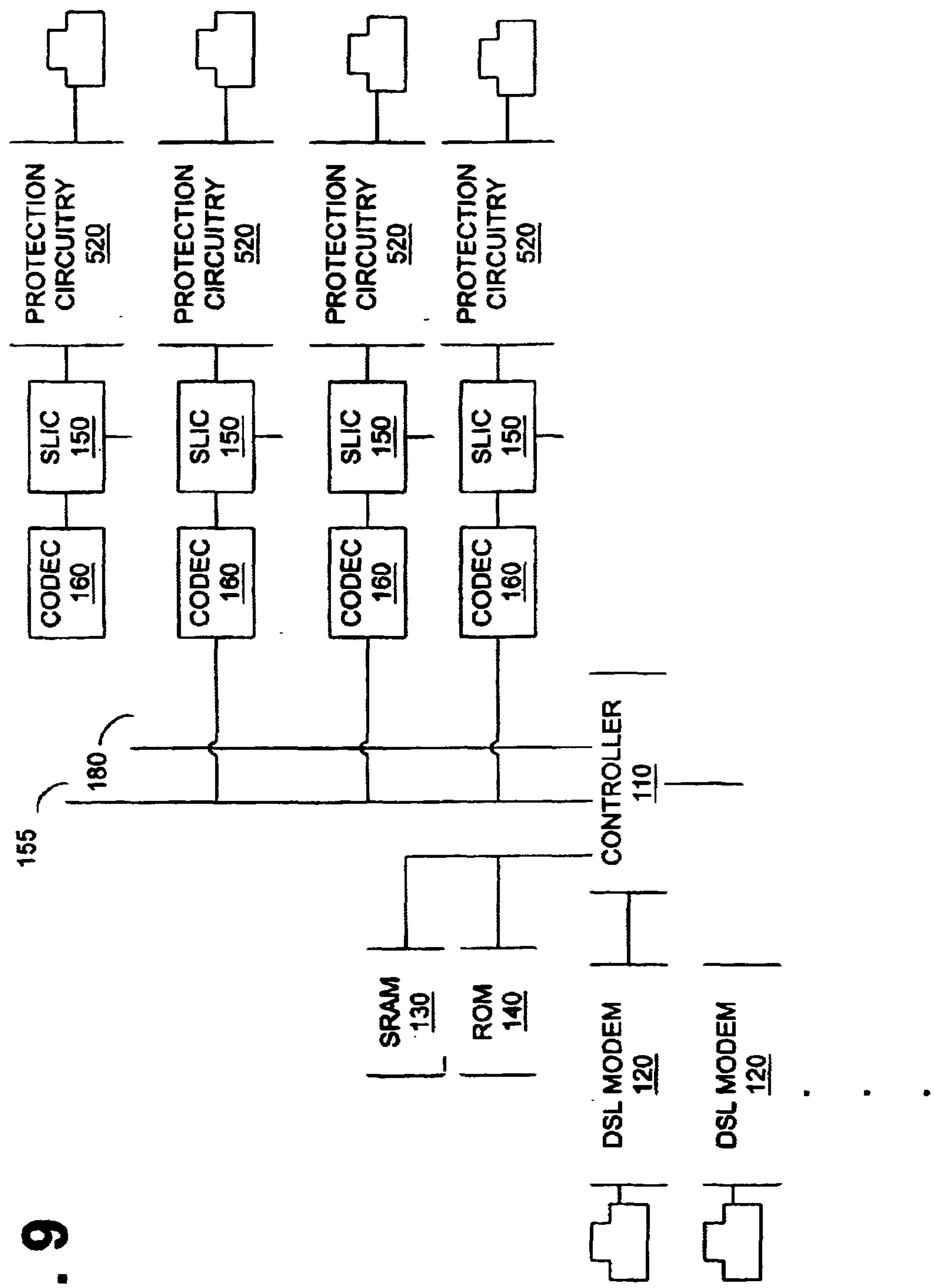
PROTECTION CIRCUITRY 520
PROTECTION CIRCUITRY 520
PROTECTION CIRCUITRY 520
PROTECTION CIRCUITRY 520
SLIC 150
SLIC 150
SLIC 150
SLIC 150
CODEC 160
CODEC 160
CODEC 160
CODEC 160
180
155
CONTROLLER 110
SRAM 130
ROM 140
DSL MODEM 120
DSL MODEM 120

FIG. 9

SYSTEM AND METHOD FOR COMMUNICATING VOICE AND DATA OVER A LOCAL PACKET NETWORK

This application is a continuation-in-part of U.S. application Ser. No. 09/112,911, filed Jul. 9, 1998, now U.S. Pat. No. 6,075,784, and claims priority to U.S. Provisional Application No. 60/088,399, filed Jun. 8, 1998.

FIELD OF THE INVENTION

The present invention is directed to a system and method for providing voice data services over an access network supporting a digital packet-based transport protocol, such as Digital Subscriber Line (DSL) technology. In addition, the present invention is directed to a system and method for providing conventional analog telephone service to supplement service provided through a digital-packet based transport network.

BACKGROUND OF THE INVENTION

Conventional or "plain old telephone signal" (POTS) telephone service is provided over an "analog loop" that carries audio communications and signaling between a customer's telephone device(s) and the telephone company's wire center A fundamental attraction of this technology is that power for the connection is provided from the wire center, and thus, with proper stand-by power systems at that location, telephone service remains available even when the customer's power is interrupted. Additionally, only a simple, and generally very reliable, telephone set is required at the customer's location, resulting in a service that is subject to minimal interruption.

Increasingly, telephone companies have used more sophisticated digital technology to deliver telephone service; this approach requires more complex equipment at the subscriber's location and that equipment is, inherently, subject to additional failure modes and requires local powering. Typically, this equipment is installed at larger business locations where it may be practical to also install stand-by power equipment and redundant electronics, so that at least some telephone service can be maintained even when equipment or conventional power fails. The telecom industry is anxious to bring the advantages of digital technology to residential subscribers, preferably without giving up the reliability associated with conventional analog phone service. It is, generally, not cost-effective to equip the residential subscriber with stand-by power equipment and redundant electronics.

DSL is a high bandwidth technology that enables data to be transferred to and from individual subscriber locations at various speeds, currently ranging as high as 2 Mbps. Data is transferred over a DSL access portion of a local packet network (LPN) as "packets," and packets move over the LPN only when information is moving to or from the subscriber, and the line is in an idle condition otherwise. An LPN is a network that provides data connections among subscribers in a local service area with various connection types and data rates. Typically, an LPN consists of a plurality of DSL multiplexers and data switches. DSL equipment is designed to serve large numbers of subscribers, resulting in relatively low per-subscriber costs.

DSL technology is a very cost-effective mechanism for delivering high-bandwidth digital data to a residential subscriber. Some variants of DSL, in particular, Asymmetric Digital Subscriber Line (ADSL) permit the superposition of the high-bandwidth digital data stream on top of traditional

analog telephone service. This is accomplished through the use of a readily available splitter/combiner device installed both at the wire center and at the subscriber's premises. Voice communications can then be provided both by the analog signal, and by additional digital channels derived from the digital stream (as explained in above). However, as noted, providing the analog voice service requires POTS-compatible equipment in the WC. Typically, this equipment is large and expensive, and requires special connections, using time-division multiplexing, to the rest of the telephone network.

The aforementioned co-pending application provides a system and methods that enable facilities-based full service Competitive Local Exchange Carriers (CLECs) to transport local telephone service, including multiple voice call service, and data services to small and medium-sized business customers over an access network that supports a digital packet-based transport protocol, preferably over existing copper wire pair lines.

A telephone company deploying the technology above may want to offer the advantages of the analog POTS capability of ADSL, but cannot cost-justify the traditional switching technology required at the wire center. They need a mechanism that leverages their investment in packet-based technology but still delivers the benefits of traditional POTS, including continuous operation even when the customer's power fails.

SUMMARY OF THE INVENTION

Briefly, the present invention is directed to a system and method for utilizing a local packet network (LPN) that supports a digital packet-based transport architecture, such as Digital Subscriber Line (DSL), to provide voice and optionally data services over a single local loop, such as a DSL, to a customer site. Multiple voice telephone calls as well as data services for a customer site are supported on a single DSL connected to that customer site.

At a customer site, a plurality of telephone devices (such as telephones, facsimile machines, modems and/or office telephone system ports) and data devices (such as those connected by a local area network) are interfaced to a local loop link connected to the LPN. Analog telephone signals (representing voice, facsimile signals, or modem signals) received from the plurality of telephone devices are converted to digital voice-band packets. Control signals representing off-hook, dial tone, call setup information, and other call control signals are converted to digital call control packets. The voice-band packets, call control packets and data packets (from the data devices at the customer site) are modulated for transmission via the local loop link over the LPN. In the reverse direction, modulated voice-band packets, data packets and call control packets received from the LPN destined for the customer site via the LPN on the local loop link are demodulated. The demodulated voice-band packets are converted to analog telephone signals for connection to appropriate ones of the plurality of telephone devices at the customer site. The demodulated data packets are coupled to the data devices (in the local area network) at the customer site. The demodulated call control packets are processed to control call setup and maintenance functions at the customer site.

At a control site within or connected to the LPN (such as a central switching facility), voice-band packets, call control packets and data packets from the customer site are received via the,LPN. The voice-band packets received from the customer site via the LPN are converted to time-division

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multiplexed signals and are coupled to a public switched telephone network (PSTN) switch in assigned time slots. Data packets received from the customer site are coupled to a data switch for transfer to a data network (such as the Internet). The call control packets are processed to control call setup and maintenance functions at the control site. In the reverse direction, data packets destined for the customer site are received from the data switch and coupled to the LPN for transmission to the customer site. Time-division multiplexed voice signals received from the PSTN switch destined for the customer site are converted to voice-band packets and are coupled to the LPN for transmission to the customer site.

A specialized apparatus called a remote digital terminal (RDT) is provided at the customer site and another specialized apparatus, called a host digital terminal (HDT) is provided at the central switching facility. Alternatively, the specialized functions of the HDT are integrated into a data switch in the LPN or into a PSTN switch. Similarly, the functions of the RDT can be integrated into a key telephone system/private branch exchange device or other equipment at the customer site. The RDT and HDT transport digitized voice-band packets and data packets between each other via the LPN. The RDT converts the voice-band packets suitable for communication over the LPN to and from analog telephone signals suitable for use by attached telephone devices. Similarly, the HDT converts voice-band packets to and from a time-division multiplexed format suitable for communication via the PSTN switch.

An apparatus called a wire center remote digital terminal (WC-RDT) is provided at a wire center to interface a plurality of local loop links with a digital subscriber line access multiplexer (DSLAM) in order to provide POTS service on the local loop links to customer sites. The WC-RDT is used to provide POTS service to customer sites having an RDT as well as to customer sites that do not have an RDT.

The above and other objects and advantages of the present invention will become more readily apparent when reference is made to the following description, taken in conjunction with the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram of a telecommunication system employing the remote digital terminal and a host digital terminal of the system and method according to the present invention.

FIG 2 is a block diagram of a remote digital terminal according to the present invention.

FIG. 3 is a block diagram of a host digital terminal according to the present invention.

FIG. 4 is a flow chart depicting a call set-up procedure in the remote digital terminal.

FIG. 5 is a flow chart depicting a call-set up procedure in the host digital terminal.

FIG. 6 is a flow chart depicting processing of voice-band data in the remote digital terminal.

FIG. 7 is a flow chart depicting processing of voice-band data in the host digital terminal.

FIG. 8 is a block diagram showing an enhancement to the system wherein a wire center remote digital terminal is provided at a wire center to provide standard analog service to customer sites without a remote digital terminal, or for customers equipped with a remote digital terminal, even if their remote digital terminal becomes inoperative.

FIG. 9 is a block diagram of the wire center remote digital terminal.

DETAILED DESCRIPTION OF THE INVENTION

FIG. 1 is a general diagram of a system 5 that enables a facilities-based full service CLEC to transport local telephone service and data services to small and medium-sized businesses. The present invention is described as being useful in utilizing a particular type of local loop link, called the DSL network. However, it should be understood that the teachings described herein are applicable to any access network supporting a digital packet-based transport protocol. DSL is only an example of such a protocol-access network technology.

The primary components of the system according to the present invention are a remote digital terminal (RDT) **100** and a host digital terminal (HDT) **200**. The RDT **100** resides at a customer (subscriber) site shown at reference numeral **7** and interfaces a plurality of telephone devices (TDs) **10** and, optionally, a local area network (LAN) **12** to a local loop link, such as a line supporting the Digital Subscriber Line (DSL) transport protocol. For simplicity, the local loop link is referred to as a DSL **24**. Other local loop links that may be suitable for use in conjunction with the present invention are wireless local loops, such as digital cellular local loops, and the like.

The DSL **24** is an access network of a local packet network (LPN) **60**. The LPN **60** comprises one or more data switches **62**, such as Asynchronous Transfer Mode (ATM) switches, and one or more DSL access multiplexers (DSLAMs) **64**. The data switches **62** consist of one or more processors controlled by software. The data switches **62** are connected to each other and to DSLAMs **64** preferably via optical links, such as synchronous optical network (SONET) facilities **66**. In each ILEC central office (CO), there is a DSLAM **64** that controls the distribution (and collection) of signals to and from a plurality of DSLs **24**. The combination of DSLAMs **64** and data switches **62** make up the LPN **60**.

The LPN **60** provides data connections among subscribers in a local service area with various connection types and data rates. For example, the LPN might include DSL connections at rates ranging from 256 Kbps to 6 Mbps used by homes and small businesses, T3 connections at 45 Mbps used by large businesses and small Internet Service Providers (ISPs) and OC-3 connections for used by the largest businesses and ISPs.

Returning to the description of a customer site **7**, the TDs **10** may connect directly to the RDT **100** or to a key telephone system/private branch exchange (KTS/PBX) device **14** that is connected to the RDT **100**. Connections between the RDT **100** and the associated TDs are by standard analog telephone lines **25**, or alternatively by other standard telephony interfaces such as T-1, ISDN, etc. Connections between the RDT **100** and the KTS/PBX **14** are by way of a plurality of trunks **27**. The function of the RDT **100** is to allow voice traffic associated with one or more TDs **10** and data traffic from the LAN **12**, if any, to be converted to and from a format that can transit the LPN **60**. It should be understood that the voice-band traffic associated with the TDs **10** could be voice, modulated digital data from a modem, facsimile machine, and possibly certain call control signals (such as dial tone, busy signal, etc.). Data traffic is that traffic associated with the LAN **12** or other data-packet based devices at the customer site **7**.

As is well known in the art, DSL is a telecommunication technology that enables data to be transferred from indi-

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vidual subscriber locations at various speeds, currently as high as 2 Mbps using the existing twisted wire pair line infrastructure already in place in most industrialized parts of the United States and the world. That is, the DSL 24 is a standard twisted wire pair line that is used to transmit information that is formatted in accordance with the DSL transport protocol.

The HDT 200 resides at control site within or without the LPN 60. For example, FIG. 1 shows the HDT residing at a CLEC switch facility 30 and interfaces a PSTN switch 32 and a data switch 34 to the fiber backbone of the LPN 60. The PSTN switch 32 may route voice calls to the local PSTN 42 or to a long distance network. The data switch 34 may route data packets to and from a data network 50, such as the Internet. The HDT 200 links via an optical fiber 26 or another facility connected to the LPN 60. The CLEC 30 switching facility is also hereinafter referred to as a central switching facility. Both the data switch 34 in the central switching facility 30 and the data network 50 may route data directly to the LPN 60, bypassing the HDT 200. Similarly, a data switch 62 in the LPN 60 may directly route data to the data network 50.

Alternatively, the functions of the HDT 200 may be incorporated into a control site at another location in the system. For example, the functions of the HDT may be incorporated into a data switch 62 of the LPN 60 rather than be performed by a separate unit. Software that carries out the functions of the HDT 200 (described hereinafter) may be provided in the data switch 62 to be executed by the processor(s) in the data switch 62. The software in the data switch 62 can be enhanced to perform the functions of the HDT 200 and interface directly to the PSTN switch 32, to the data switch 34 or to the data network 50. Similarly, the software may be provided in a PSTN switch 32 to be executed by processors associated with the PSTN switch 32.

Turning to FIG. 2, with continued reference to FIG. 1, the components of the RDT 100 will be described. In a preferred embodiment, the RDT 100 is embodied as circuit board housed in a suitable enclosure with a power supply.

Specifically, according to a preferred embodiment, the RDT 100 comprises a controller 110, a DSL modem 120, a static random access memory (SRAM) 130 for buffering working data, a read only memory (ROM) 140 that stores a software program for the controller 110, a plurality of subscriber line interface circuits (SLICs) 150, a plurality of coder/decoders (CODECs) 160, and an Ethernet interface 170.

The DSL modem 120 connects directly to the DSL 24. The DSL modem performs the modulation and demodulation necessary to transport information via the DSL 24 into the LPN 60. There are several modulation/demodulation formats that are known in the art for use over a DSL 24. The DSL modem 120 also formats the modulated information into a suitable packet format, such as the asynchronous transfer mode (ATM) protocol for example, that is utilized by the equipment in the LPN 60 for the transport of information. Alternatively, if the local loop link is a wireless local loop link, the modem 120 would be a wireless modem capable of performing the modulation and demodulation necessary for transporting information via a wireless link. In addition, a transceiver (not shown) would be connected to the modem 120 to wirelessly transmit and receive the modulated information.

The controller 110 is connected to the DSL modem 120, the SRAM 130, ROM 140, Ethernet interface 170 and a time division multiplex (TDM) bus 180. Furthermore, the controller is connected to each of the SLICs 150 by a control bus 155 to enable the controller 110 to detect when a TD 10 goes off-hook and to command an SLIC 150 to ring a TD 10. Each SLIC 150 is connected to a TD 10 or to a PBX/KTS 14 by a standard analog telephone line. The SLIC 150 provides the precise voltages and currents required to interface to telephone devices, such as standard telephone sets, facsimile machines, etc. The CODEC 160 is a coder/decoder that converts analog telephone signals (voice and other in-band telephone signals) to digital bit streams, and converts digital bit streams to analog telephone signals. The controller 110 is connected to each of the CODECs 160 by the TDM bus 180. The number of TDs 10 serviced by the RDT 100 will determine the number of SLICs 150 and CODECs 160 required. However, as will be explained hereinafter, the number of telephone devices that may be in use at any one time depends on the bandwidth of the local loop link. The Ethernet interface 170 is a standard network interface circuit device that interfaces digital data between an attached PC or LAN 12.

The controller 110 is preferably a microprocessor that operates in accordance with software stored in the ROM 140. The operation of the controller 110 may be updated or modified by employing a reprogrammable non-volatile ROM, such as a "flash" memory, that is well known in the art. The controller 110, under control of the software program stored in the ROM 140, performs two major functions: call set-up control and voice-band data conversion.

As an alternative to the implementation shown in FIG. 2, the RDT 100 may be implemented using a PC with plug-in cards that provide the necessary interfaces (DSL, telephone, and Ethernet). Still another alternative is to provide a server computer that provides the call control functionality to a plurality of multimedia client PCs each having plug-in telephony and sound cards so that each PC can support a telephone call. Yet another alternative is to implement the functions of the controller 110 by a digital signal processor (DSP) or an application specific integrated circuit (ASIC).

Furthermore, rather than providing a plurality of individual SLICs 150, a single subscriber line interface unit capable of coupling a plurality of signals to and from a plurality of telephone devices may be employed. Similarly, a single voice conversion device having the processing capability to perform multiple conversions may be used in place of the plurality of separate CODECs 160.

The function of the RDT 100 is to allow traffic associated with one or more telephone calls to be converted to and from a form that can transit the LPN 60. More specifically, the functions of the RDT 100 include interfacing to a local loop link, such as a DSL (via an integrated or external DSL modem); converting voice-band packets to and from conventional analog telephone signals; converting data packets to and from a format suitable for transport via the LPN; processing call control functions (ringing, on-hook, and off-hook functions) to generate and detect call control packets sent or received via the LPN; providing an electrical interface to conventional telephone equipment; managing the sharing/allocation of the bandwidth on the local loop link with other (non-telephony) functions; and providing remote management and maintenance functions.

Turning to FIG. 3, in conjunction with FIG. 1, the HDT 200 will be described. The HDT 200 performs functions complementary to those performed by the RDT 100. A single HDT 200 can support communication with a plurality of RDTs 100 that are connected to the LPN 60.

In the preferred embodiment, the HDT 200 is implemented in a conventional computer system, with specialized software controlling a set of interface electronics. Specifically, the HDT **200** shown in FIG. **3** comprises a microprocessor-based controller **210**, a bus, such as a Peripheral Connection Interface (PCI) bus **220**, a storage disk **230**, a display **240**, and a memory **250**, such as SRAM. The HDT **200** interfaces to the PSTN switch **32** by a PSTN interface (IF) device, such as T-1 IFs **260**, preferably communicating via the Bellcore GR-303 signaling interface. That is, the T-1 IFs **260** connect to T-1 lines, which are in turn connected to the PSTN switch **32**. The HDT **200** interfaces to an optical link connected to the LPN **60** by an LPN IF device **270**, such as an OC-3 optical link utilizing the Asynchronous Transfer Mode (ATM). Similarly, a data switch IF device **290** is provided to interface with the data switch **34**. PC-cards implementing the functions of the LPN IF device **270** and data switch IF **290** are plugged into the PCI bus **220**.

The LPN IF device **270** connects to the LPN **60** via an optical fiber (FIG. **1**) and provides a data transfer rate of 155 Mbps in each direction. ATM cells are sent and received over the link, wherein each cell contains address information, including source and destination information, as well as the data to be transferred. Alternatively, the LPN IF device **270** operates at other rates, such as OC-12 at 622 Mbps, Fast Ethernet (100 Mbps) or Gigabit Ethernet (1000 Mbps.) However, the preferred embodiment is lo the ATM-155 standard.

The T-1 IFs **260** are coupled to a packet conversion processor **280** by a TDM bus **285**. Similarly, the data switch IF **290** and LPN IF **270** are coupled to the packet conversion processor **280** by a TDM bus **287**. The function of the packet conversion processor **280** is to convert information between different formats used by the devices connected to it. For example, the packet conversion processor **280** converts packets received from the LPN via the LPN IF **270** to a time-divisional multiplexed format for coupling to the T-1 IFs **260**. Similarly, the packet conversion processor **280** converts packets received from the LPN IF **270** to a suitable format for coupling to the data switch IF **290**. The packet conversion processor **280** performs these conversions in reverse as is appropriate. The packet conversion processor **280** is preferably implemented by an application-specific processor, and its operation is supervised by the microprocessor/controller **210** for call control functions, call setup, system errors and other matters. In some configurations, the function of the packet conversion processor **280** may be included within the functionality of the microprocessor **210**.

The T-1 facilities connected to the PSTN switch **32** are logically divided into "time-slots" using time-division multiplexing. Each T-1 line carries 24 time-slots or channels, with each channel carrying a single digitized voice conversation. One or more of the 24 channels is designated a control channel and carries signaling information between the HDT **200** and the attached PSTN switch **32**. Use of the control channel is described in more detail hereinafter.

Preferably, the HDT **200** is designed to have a modular architecture that is easily scaleable. The HDT **200** can cost-effectively support as few as 100 end user lines and 24 trunks to the voice switch **32**, and up to as many as 6,000 lines and 2,000 trunks. Additional HDTs can be added to handle as many as 100,000 lines or more. The HDT **200** is preferably designed to provide "carrier-class" availability, including redundant and "hot-swappable" components. It is preferably a NEBS level 3 compliant rack-mounted system designed to reside in a central office environment and can be engineered to have N+1 redundancy.

A software program stored in the memory **250** allows the microprocessor **210** to perform the control functions analogous to the call set-up and functions performed by the RDT **100**, and to support communication with multiple RDTs **100** simultaneously. As mentioned above, software to carry out the functions of the HDT **200** may be incorporated directly into the PSTN switch **32** or into the data switch **62** such that a separate "box" to carry out the functions of the HDT **200** would not be needed.

The operations of the RDT **100** and HDT **200** are described with reference to FIGS. **4–7**, together with FIGS. **1–3**. In order to transport voice and data services via the DSL access portion through the LPN **60**, the RDT **100** and HDT **200** employ a compatible digital signaling and information transfer protocol. There are many such protocols well known in the art of telecommunications, and it is likely that many new protocols will be created that may be useful in connection with the system and method of the present invention. The ATM protocol is the example of a suitable protocol. The ATM protocol is a format that divides a bandwidth into a plurality of cells each of which may contain voice-band packets, data packets, call control packets, etc.

The local loop link supports in each direction the transport of voice-band packets representing analog voice-band telephone signals, data packets associated with data devices (computers in a LAN) and call control packets. A voice-band packet includes an identifier (source and destination) and the voice-band packet information. A call control packet includes a control flag (identifying it as a control message), a control message (off-hook, busy, etc.) and an identifier associated with the call to which the control message applies. If the ATM protocol is used, each packet occupies an ATM cell. Upon receiving a packet, the RDT **100** and HDT **200** first detect the type of packet received (by a control flag or other identifying information) in order to determine whether the packet represents call control functions, voice-band information or data information. The RDT **100** converts analog telephone signals to and from a suitable digital packet format in order to communicate with the attached analog TDs **10**. In addition, if communication with data devices at the customer site is supported and required, the RDT **100** converts data packets from the data devices to and from a suitable packet format for communication via the LPN. Similarly, the HDT **200** converts voice-band information and data packets between different types of digital formats in order to conduct communications via the PSTN switch **32** and the data switch **34**.

Turning to FIG. **4**, with reference to FIG. **2**, a call-setup procedure **300** in the RDT **100** is described. This procedure occurs when a call is initiated by one of the TDs **10** connected to the RDT **100**. In step **310**, an SLIC **150** detects that a connected TD **10** is off-hook, and a corresponding signal is coupled to the controller **110**. The controller **110**, in step **315**, generates a control message (formatted into a call control packet) indicating the off-hook status and requesting a dial-tone, that is transmitted to the HDT **200** via the LPN **60**. The controller **110**, in doing this, first determines whether there is available bandwidth on the DSL **24** based on the number of other voice calls currently being maintained by the RDT **100**. If there is available bandwidth, then the message is sent to the HDT **200**; otherwise, a "busy" or other signal indicating unavailability is sent to that telephone device.

Next, in step **320**, the RDT **100** receives a call control packet from the HDT **200** that includes a call reference value to be associated with that call. In step **325**, a call control

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packet or voice-band packet (depending on the system implementation) representing a voice-band dial-tone is received from the HDT **200** that is converted to a digital bit stream by the RDT **100**, converted to an analog dial-tone signal by the CODEC **160** and connected to the TD **10** by the SLIC **150**.

FIG. **5** illustrates the complementary call set-up procedure **400** in the HDT **200**, again, in the case when a call is initiated by a telephone device connected to the RDT **100**. The use of the ATM protocol is referred to in the following description as an example of a suitable networking technology used by the LPN. Reference is also made to FIG. **3** in connection with this description. In step **410**, the HDT **200** receives the call control packet containing a control message from the RDT **100** indicating that a telephone device has gone off-hook. In step **415**, the HDT **200** identifies the source of the call control message by looking at the address provided in the ATM cell containing the call control packet request. It is verified whether the request is valid, i.e., coming from a customer site **7** whose account is active. In step **420**, the HDT **200** notifies the PSTN switch **32** via the designated control channel over one of the T-1 IFs **260** of the request. In reply, the PSTN switch **32** assigns an available time-slot for the call on one of the T-1 facilities, and this time-slot information is received by the HDT **200** in step **425**, together with a dial-tone over that time-slot. Next, in step **430**, the HDT **200** assigns a call reference value for the call and communicates that value in a control message that is transmitted to the RDT **100**. Finally, in step **435**, the HDT **200** generates voice-band packet(s) representing a dial-tone for the call and transmits it to the RDT **100**.

Calls that are initiated by devices on the HDT side of the system are received by the HDT **200** when the PSTN switch **32** detects a match between a received telephone number and a table of telephone numbers assigned to a customer site within the supervisory control of the HDT **200**. In response, the HDT **200** transmits to the RDT **100** a call control packet containing a command for a ringing signal that is interpreted by the RDT **100** to generate an analog ringing signal for connection to the appropriate TD **10**. A call reference value is assigned, and voice-band packets are exchanged between the HDT **200** and RDT **100** in a manner analogous to that described for a call initiated at the RDT **100**. When the addressed telephony device at the RDT **100** goes off-hook, the RDT **100** generates a control packet indicating same, and the HDT **200**, upon receipt of that packet, notifies the PSTN switch **32** that the call has been answered. Thereafter, the operation of the RDT **100** and HDT **200** continues as explained below. The communication of a data packet to a data device at the customer site is initiated in an analogous manner when the data switch **34** receives a data packet destined for a data device at the customer site.

It should be understood that the "telephone numbers" associated with the TDs **10** connected to the RDT **100** are managed by the PSTN switch **32** located in the central switching facility **30**. When a person somewhere in the PSTN **42** dials a number, such as 404-555-1234, traditional PSTN technology routes that call to the PSTN switch **32**, which in turn presents the call information, as described herein, to the HDT **200**, for transmission to the RDT **100** and ultimately to the connected TD **10**. Thus, the TDs **10** connected to the RDT **100** operate identically to telephone devices connected directly to the central switching facility **30**, and can make and receive PSTN telephone calls in the traditional fashion.

Once a call between the RDT **100** and HDT **200** is set up, remaining communication will consist primarily of voice-

band packets or data packets depending on the end devices communicating with each other until the call is terminated. In general, only when new calls are initiated, calls are terminated, or system errors occur will call control packets occur over the local loop link.

The operation of the RDT **100** once a call is set-up is described with reference to FIG. **6**, in conjunction with FIGS. **1** and **2**. There are two major conversion processes that occur at the RDT **100**: analog-to-digital and digital-to-analog conversion shown at step **510**, and digital bit stream-to-packet and packet-to-bit stream conversion at step **520**. The analog-to-digital (and vice versa) conversion is performed by the CODECs **160** for the analog telephone signals. Some call control related telephone signals are generated by the controller **110** via the SLICs **150**. The bit stream-to-packet (and vice versa) conversion is performed by the controller **110**.

To explain further, the flow of signals from a TD **10** to the LPN **60** is first described. Analog telephone signals from an attached TD **10** are received and digitized by the attached CODEC **160**, creating a digital bit stream representing the real-time analog telephone signals generated by the TD **10**. The digital bit stream is placed on the TDM bus **180**. In step **520**, the digital bit stream is converted into digital packets called voice-band packets. Each voice-band packet contains a plurality of bytes of voice-band information, each representing a sample of the speech (or analog telephone signals) at a predetermined time interval, such as (125) microsec. A single voice-band packet contains a predetermined number of samples of voice information, such as 40, representing 5 msec. of speech, for example. More specifically, the controller **110** accepts 1 byte of digital data from the TDM bus **180** each 125 microsec., and buffers the information in the SRAM **130**. When a full packet of voice information is accumulated in the SRAM **130**, the controller **110** formats the packet with its associated call reference value and transfers it to the DSL modem **120**, causing it to be modulated and transmitted via the LPN **60** to the HDT **200**. The controller **110**, under control of the software stored in the ROM **140**, carries out this process for all calls that are active at the RDT **100**. Similarly, when the controller **110** receives data packets via the Ethernet interface **170** from the LAN **12**, it buffers them in the SRAM **130** for modulation by the DSL modem **120** and transmission via the LPN **60**. Depending on the availability of bandwidth on the DSL **24**, the controller **110** may buffer the digital data from the LAN **12** until sufficient bandwidth becomes available, giving priority to the voice-band information.

The controller **110** carries out this process for each of the TDs **10** which is active at any given time, sequencing through each CODEC **160**, accumulating a voice-band information for each call, and passing it to the DSL modem **120** for transmission when it is filled. Construction of the packets is skewed by the controller **110** such that all packets are not ready for transmission simultaneously; rather, a packet for a first call is completed, and then the packet for the next call is completed a fraction of a second later, allowing the packet for the first call to be transmitted by the DSL modem **120** so that it is ready to accept the packet for the next call when it is ready. Alternatively, the voice-band data for several calls may be multiplexed together into a single voice-band packet that contains identifiers to each.call and data for each call. This alternative technique reduces any potential delay that may occur as a result of the packetizing process.

In the reverse direction, the DSL modem **120** receives modulated voice-band packets and data packets transmitted

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over the DSL **24** from the central switching facility **30**. The DSL modem **120** demodulates the modulated voice-band packets and modulated data packets and couples them to the controller **110**. The controller **110** receives the demodulated voice-band packets and data packets, and identifies voice-band packets by the call reference value contained therein. The controller **110** stages the voice-band packets in SRAM **130**, queuing subsequent voice-band packets as they are received. Simultaneously, the controller **110** takes individual bytes of data from the SRAM **130** and places them onto the TDM bus **180** at the rate of 1 byte every 125 microsec. for connection to an appropriate one of the CODECs **160** associated with the addressed TD **10**. The CODEC **160** converts the digital bit stream data to analog telephone signals, which are coupled by the SLIC **150** to the TD **10**.

With respect to the received data packets, when the controller **110** detects the reception from the DSL modem **120** of a data packet not associated with telephony functions, it directs it to the Ethernet interface **170** where it is coupled (according to an associated address) to the appropriate data device in the LAN **12**.

The RDT **100** sets the priority for utilization of the bandwidth of the local loop link between voice traffic and data traffic. For example, the controller **110** may be programmed to assign priority of use of the local loop link to voice traffic over data traffic. In this case, bandwidth over the DSL connection will be used for voice calls when they are active, but will be available for data traffic when some of the TDs **10** are idle. Typically, the bandwidth of a DSL is as high as 2 Mbps. The controller **110** may designate a portion of that bandwidth for voice traffic, for all or adjustable time periods during a day. Once the data traffic maximum is reached, no further data traffic would be permitted. These traffic parameters are programmable in the RDT **100**, and if necessary, can be adjusted in real-time to accommodate sudden bandwidth allocation needs.

With reference to FIG. **7**, in conjunction with FIGS. **1** and **3**, the operation of the HDT **200** in processing ongoing calls is described. Voice-band packets and data packets sent by the RDT **100** via the LPN **60** are received by the LPN IF **270**. The packet conversion processor **280** converts the voice-band packets to time-division multiplexed (TDM) signals in a process similar to that in the RDT **100**, and couples the TDM signals via a T-1 IF **260** to the PSTN switch **32** in an assigned time-slot. Data packets received from the RDT **100** are (reformatted if required by the packet conversion processor **280** and) coupled to the data switch **34** by the data switch IF **290** under control of the microprocessor **210**.

In the reverse direction, the HDT **200** receives TDM signals for a given call from a T-1 IF **260** at a rate of one byte every 125 microsec. In a process similar to that in the RDT **100**, the packet conversion processor **280** buffers the bytes of data of the TDM signals in the memory **250** to form voice-band packets, which are then dispatched to the RDT **100** via the LPN IF device **270**. Data packets received from the data switch **34** are coupled to the LPN IF device **270** for transmission to the customer site.

The HDT **200** may also be involved in bandwidth usage control. For example, the microprocessor **210** may be programmed to continuously monitor the amount of bandwidth of the local loop link used by a given RDT **100** at customer site for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic. The microprocessor **210** may be programmed to limit data traffic to the customer site from the LPN such that a predetermined portion of the total bandwidth available to the customer site on the local loop link used by that RDT **100** remains available for voice traffic. Furthermore, the microprocessor **210** may be programmed to generate a message to be sent to a data switch **62** in the LPN **60** indicating the amount of bandwidth available for data so that the data switches **62** in the LPN **60** can abide by this rule in transmitting (or not transmitting) data traffic to the HDT **200** destined for a particular RDT **100** at a particular customer site.

In order to be competitive in the local market, CLECs must deliver service that is of the same quality as that which the customer is currently receiving from the ILEC. To achieve this, the system of the present invention digitizes voice conversations using standard "μ-law" encoding (64 Kbps) and need not use voice compression (though compression can be used if it is desired to reduce the bandwidth utilization through the LPN). In addition, to avoid the delays (latency) inherent in some packet networks, the system according to the preferred embodiment uses ATM signaling and is optimized throughout to limit buffer sizes and queues, insuring that voice-band data moves expeditiously between the RDT and HDT.

Turning to FIGS. **8** and **9**, the wire center remote digital terminal (WC-RDT) is described. The WC-RDT is similar to the RDT, with some modifications, and is located at the wire center (WC) shown, for example, at reference numeral **68** in FIG. **8**. The WC-RDT **500** connects to a multiplexer in the WC **68**, such as a DSLAM **64** and provides POTS interfaces for a large number of subscribers.

The WC-RDT **500** is shown in FIG. **9**. It is similar to the RDT **100** shown in FIG. **2**, but lacks the Ethernet port **170**. The WC-RDT **500** comprises a plurality of subscriber line interface circuits **150** each suitable for connection to the local loop links of a plurality of customer sites via one of the plurality of standard telephone (POTS) ports **510**, and a plurality coder/decoders **160** each for connection to a subscriber line interface circuit **150**. Each coder/decoder **160** is capable of converting analog telephone signals received from an associated subscriber line interface circuit **150** to digital bit streams and converting a digital bit stream to analog telephone signals to be supplied to the associated subscriber line interface circuit.

A controller **110** is coupled to the coder/decoders **160** and to the subscriber line interface circuits **150**. The controller **110** manages the interface of voice-band packets between the DSLAM **64** and the plurality of POTS ports **510**. Multiple DSL modems **120** may be provided in the WC-RDT **500** to handle higher capacity through multiple connections to the DSLAM **64**. The WC-RDT **500** is very compact, and can be implemented in a rack-mounted form factor such that very high port densities can be achieved. Alternatively, the functions of the WC-RDT **500** can be integrated into a DSLAM **64**.

Likewise, the operation of the WC-RDT **500** is similar to that of the RDT **100**. The controller **110** managing the interface of voice-band packets between the DSLAM **64** and the plurality of standard telephone ports by: converting digital bit streams received from one or more coder/decoders **160** to voice-band packets and coupling the voice-band packets to the LPN for transmission to the host digital terminal **200**; and converting demodulated voice-band packets received by the DSLAM **64** to digital bit streams and coupling the digital bit streams to an appropriate one of the coder/decoders **160**.

The WC-RDT **500** is powered by the power facilities of the wire center **68** where the DSLAM **64** is located. These facilities are generally highly reliable, and include batteries and back-up generators to assure operation even during commercial power failures. The WC-RDT **500** need not be equipped with connectivity for a LAN, unlike the RDT **100**. The wire center **68** may be a large neighborhood building that houses switching systems and other communications equipment, or could be a cabinet or vault that houses a DSLAM **64** that provides connections to subscriber loops and includes a high-availability power system.

The WC-RDT **500** is connected to a DSLAM **64** by any available technology supported by the DSLAM, including the same technology used by the RDT **100**. In fact, DSL technologies are somewhat sensitive to distance, and therefore tend to provide higher bandwidth capacity when the distance between two devices is shorter. When both the WC-RDT **500** and the DSLAM **64** are located in the same room of a wire center facility, the DSL technology can be expected to perform at its highest capacity. For example, using available and relatively inexpensive symmetric DSL technology, the bandwidth can be as high as 2.048 Mb/sec in each direction. More than one link can be provided between a WC-RDT **500** and a DSLAM, thereby providing additional bandwidth in the connection. The WC-RDT **500** can use this additional bandwidth to serve more POTS ports or lines to more subscribers. In addition, multiple links between the WC-RDT **500** and the DSLAM **64** provide redundancy to maintain the reliability of the network.

Referring back to FIG. **8**, the WC-RDT **500** communicates with the HDT **200** in the central switching facility **30**. Telephone service for each POTS port **510** of the WC-RDT is provided by the HDT **200** according to the operations described above with reference to FIGS. **3** and **4**. No additional transport or switching facilities are needed. The HDT **200** communicates with the WC-RDT **500** in the same manner that it does with the RDT **100**. However, the WC-RDT provides POTS-like service on the ports **510** such that from the perspective of a subscriber who connects a TD **10** to a port **510**, the port **510** acts just like any POTS port. Still, a subscriber who also has a RDT **100** can take advantage of the bandwidth provided to have multiple voice lines and data connections, and have back-up POTS service on at least one line that will survive a failure of the RDT **100** for power outages, etc.

The primary purpose of the WC-RDT **500** is to provide POTS service (analog telephone signals) to a plurality of TDs **10** located anywhere in the coverage area of a DSLAM **64**. The CLEC has several service delivery options. First, to deliver conventional POTS service to a subscriber, the telephone company or CLEC connects a TD **10** directly to a POTS port **510** on the WC-RDT **500**, as shown in FIG. **8**.

Second, to deliver DSL data service and a single line of POTS service to a subscriber site such as the one shown at reference numeral **700**, a subscriber's loop is connected to a splitter/combiner **600** at the wire center **68**, which splits the signal from the subscriber site into two instances, one of which is connected to a POTS port **510** of the WC-RDT and the other is connected to a subscriber port of the DSLAM **64**. At the subscriber site **700**, another splitter/combiner **610** is provided that connects to the subscriber's loop and splits it into two instances to connect to a TD **10** to provide POTS service thereto, and to a DSL modem **702**. The DSL modem **702** may connect to a PC **704** and/or a LAN **706**. Thus, the subscriber site **700** receives conventional POTS service to a standard analog TD **10**, and DSL data service to the PC **704** or LAN **706**.

Third, to deliver multi-line voice service, POTS service and DSL data service to a subscriber site such as the one shown at reference numeral **800**, the subscriber's loop is connected to a splitter/combiner **620** at the wire center **68**. The splitter/combiner **620** splits the signal from the subscriber site into two instances, one of which his connected to a POTS port **510** of the WC-RDT **500** and the other is connected to a subscriber port of the DSLAM **64**. Likewise, at the subscriber site **800**, a splitter/combiner **630** is provided to split the signal into two instances, one of which is connected to the RDT **100**, and the other is connected to a standard analog TD **10'**. Therefore, at the subscriber site **800**, the TD **10'** receives POTS service, whereas the TDs **10** receive voice service through the RDT **100**. Moreover, DSL data service is provided to the LAN **12** or PC via the RDT **100**.

The WC-RDT **500** is scaled to provide many POTS connections. It is unlikely that all of the subscribers using TDs **10** and **10'** (connected to the WC-RDT **500** via ports **510**) will use their POTS connections simultaneously. Therefore, the capacity of the DSL link between the DSLAM **64** and the WC-RDT **500** can be oversubscribed. For example, if each DSL link can support **16** simultaneous voice conversations, and the WC-RDT **500** services two DSL links between it and the DSLAM **64**, then the CLEC may choose to connect 128 POTS ports on the WC-RDT **500**, knowing that almost never will more than one-quarter of the subscribers attempt to use their POTS connections at the same time. As the number of subscribers in any region serviced by a particular wire center (DSLAM or group of DSLAMs) increases, additional WC-RDTs **500** can be installed.

Furthermore, each POTS port **510** on the WC-RDT **500** is equipped with protection circuitry **320** to provide lightening and over-voltage protection. Protection circuitry of this type is well known in the art. This is in contrast to the RDT **100** which does not include such protection circuitry on its POTS ports at the SLICs **150** because the POTS-like phone services provided by the RDT **100** are provided locally within the same building. However, connections to the WC-RDT **500** are made over "outside plant" facilities (standard telephone lines). These lines may be struck by lightening or otherwise shorted to a high-voltage supply. Therefore, to protect the circuitry of the WC-RDT **500**, suitable protection circuitry is added between the SLICs **150** and the associated connecting POTS port **510** (FIG. **2**).

In summary, the present invention is directed to a method for communicating voice to and from at least one customer site over a local packet network (LPN) supporting a packet-based transport protocol. At a customer site, a plurality of telephone devices are interfaced to a local loop link, such as a DSL or wireless local loop, connected to the LPN. Analog telephone signals received from the plurality of telephone devices are converted to voice-band packets and the voice-band packets are modulated for transmission via the local loop link over the LPN. In the reverse direction, modulated voice-band packets received from the LPN on the local loop link are demodulated. The demodulated voice-band packets are converted to analog telephone signals for connection to appropriate ones of the plurality of telephone devices. In addition to the communication of voice, the method further supports the communication of data to and from data devices (which data devices may be part of a local area network) at the customer site. Data packets from the data devices are modulated and transmitted via the LPN. In the reverse direction, data packets received from the LPN are demodulated and coupled to the data devices (in the local area network) at the customer site.

At a control site within the LPN (such as a.data switch in the LPN) or connected to the LPN (such as a HDT or a

PSTN switch at a central switching facility), voice-band packets (and optionally data packets) from the customer site are received (via the LPN if the control site is external to the LPN). The voice-band packets received from the customer site via the LPN are converted to time-division multiplexed signals and are coupled to a public switched telephone network (PSTN) voice switch in assigned time slots. Data packets received from the customer site are (reformatted if necessary and) coupled via a data switch to a destination data network, such as data network. In the reverse direction, data packets destined for the customer site are received from a source data network via the data switch connected to the source data network, such as data network. Time-division multiplexed signals received from the PSTN switch destined for the customer site are converted to voice-band packets and are coupled to the LPN **60** for transmission to the customer site. Similarly, data packets received from the source data network via the data switch are coupled to the LPN for transmission to the customer site.

Furthermore, at a wire center in or connected to the LPN, POTS service is provided to a customer site by connecting to one or more subscriber line ports of a multiplexer at the wire center; providing one or more standard telephone ports for connection to the local loop links of one or more customer sites; demodulating modulated voice-band packets transmitted from the control site via the LPN and received at the one or more subscriber line ports of the multiplexer; converting the demodulated voice-band packets to analog telephone signals for connection to the one or more local loop links of the one or more customer sites; converting analog telephone signals received on the local loop links from the one or more customer sites to voice-band packets and modulating the voice-band packets; and coupling the modulated voice-band packets to the one or more subscriber line ports of the multiplexer for transmission over the LPN to the control site

Another method according to the present invention involves communicating voice over a local packet network (LPN) to and from at least one customer site connected to the LPN via a local loop link. At a wire center in or connected to the LPN, the following steps are performed: connecting to one or more subscriber line ports of a multiplexer located at the wire center; providing one or more standard telephone ports for connection to the local loop links of one or more customer sites; demodulating modulated voice-band packets received from at the one or more subscriber line ports of the multiplexer; converting the demodulated voice-band packets to analog telephone signals for connection to the one or more local loop links of the one or more customer sites; converting analog telephone signals received on the local loop links from the one or more customer sites to voice-band packets and modulating the voice-band packets; and coupling the modulated voice-band packets to the one or more subscriber line ports of the multiplexer for transmission over the LPN to the control site. At a control site within or connected to the LPN, the following steps are performed: receiving voice-band packets from the wire center via the LPN; converting voice-band packets received from the wire center to time-division multiplexed signals and coupling the time-division multiplexed signals to a public switched telephone network (PSTN) switch; and converting time-division multiplexed signals received from the PSTN switch destined for the wire center to voice-band packets and coupling the voice-band packets to the LPN for transmission to the wire center.

Similarly, the present invention is directed to a system for communicating voice and data over an LPN to and from at least one customer site (usually a plurality of customer sites) connected to the LPN via a local loop link, comprising:

a remote digital terminal at the customer site, the remote digital terminal interfacing a plurality of telephone devices at the customer site to the local loop link to transmit and receive voice via the LPN; and a host digital terminal at a control site within or connected to the LPN that interfaces voice calls between the remote digital terminal and a public switched telephone network (PSTN) switch via the LPN.

The system may support the further transport of data between data devices (which may be part of a local area network) at the customer site and a data network within the LPN or a data network connected to the host digital terminal. Suitable hardware and the associated control functions required in the remote digital terminal and host digital terminal for transporting voice traffic and data traffic to and from a customer site are described above.

Further, in this system, a wire center remote digital terminal is provided. The wire center remote digital terminal connects to or is incorporated in a digital subscriber line access multiplexer (DSLAM) that is connected to or part of the LPN. The wire center remote digital terminal provides standard analog telephone service between a plurality of standard telephone ports suitable for connection to local loop links of a plurality of customer sites and the PSTN switch through the host digital terminal. The wire center remote digital terminal comprises: a plurality of subscriber line interface circuits each suitable for connection to the local loop links of a plurality of customer sites via one of the plurality of standard telephone ports; a plurality of coder/decoders each for connection to a subscriber line interface circuit, each coder/decoder capable of converting analog telephone signals received from an associated subscriber line interface circuit to digital bit streams and converting a digital bit stream to analog telephone signals to be supplied to the associated subscriber line interface circuit; and a controller coupled to the coder/decoders and to the subscriber line interface circuits. The controller manages the interface of voice-band packets between the DSLAM and the plurality of standard telephone ports by: converting digital bit streams received from one or more coder/decoders to voice-band packets and coupling the voice-band packets to the LPN for transmission to the host digital terminal; and converting demodulated voice-band packets received by the DSLAM to digital bit streams and coupling the digital bit streams to an appropriate, one of the coder/decoders.

Another system according to the present invention communicates voice over a LPN to and from at least one customer site connected to the LPN via a local loop link, wherein the a remote digital terminal is not necessarily located at the customer site. This system comprises a host digital terminal at a control site within or connected to the LPN that interfaces voice calls between the LPN and a public switched telephone network (PSTN) switch via the LPN; and a wire center remote digital terminal that connects to or is incorporated in a digital subscriber line access multiplexer (DSLAM) that is connected to the LPN, the wire center remote digital terminal providing standard analog telephone service between a plurality of standard telephone ports suitable for connection to local loop links of a plurality of subscribers and the PSTN switch through the host digital terminal.

In this system, the wire center remote digital terminal comprises: a plurality of subscriber line interface circuits each suitable for connection to a telephone device via one of the plurality of standard telephone ports; a plurality of

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coder/decoders each for connection to a subscriber line interface circuit, each coder/decoder capable of converting analog telephone signals received from an associated subscriber line interface circuit to digital bit streams and converting a digital bit stream to analog telephone signals to be supplied to the associated subscriber line interface circuit; and a controller coupled to the coder/decoders and to the subscriber line interface circuits. The controller manages the interface of voice-band packets between the DSLAM and the plurality of standard telephone ports by: converting digital bit streams received from one or more coder/decoders to voice-band packets and coupling the voice-band packets to the LPN for transmission to the host digital terminal; and converting demodulated voice-band packets received by the DSLAM to digital bit streams and coupling the digital bit streams to an appropriate one of the coder/decoders.

The above description is intended by way of example only and is not intended to limit the present invention in any way except as set forth in the following claims.

What is claimed is:

1. A system for communicating voice over a local packet network (LPN) to and from a customer site connected to the LPN via a local loop link, comprising:

a remote digital terminal at the customer site, the remote digital terminal interfacing a plurality of telephone devices at the customer site to the local loop link to transmit and receive voice via the LPN;

a host digital terminal at a control site within or connected to the LPN that interfaces at least one voice call between the remote digital terminal and a public switched telephone network (PSTN) switch via the LPN; and

a wire center remote digital terminal that connects to a digital subscriber line access multiplexer (DSLAM), the wire center remote digital terminal in communication with the LPN, the wire center remote digital terminal providing a standard analog telephone service between each of a plurality of standard telephone ports and the PSTN switch through the host digital terminal, where each of the plurality of standard telephone ports is suitable for connection to a subscriber loop link, where the subscriber loop link is communicatively coupled with a subscriber site.

2. The system of claim 1, wherein the wire center remote digital terminal comprises:

a plurality of subscriber line interface circuits, each of the plurality of subscriber line interface circuits suitable for connection to the subscriber loop link via one of the plurality of standard telephone ports;

a plurality of coder/decoders, each for connection to one of the plurality of subscriber line interface circuits, each of the plurality of coder/decoders capable of converting a first analog telephone signal received from the corresponding one of the plurality of subscriber line interface circuits to a first digital bit stream and converting a second digital bit stream to a second analog telephone signal to be supplied to one of the plurality of subscriber line interface circuits;

a controller coupled to the plurality of coder/decoders and to the plurality of subscriber line interface circuits, the controller managing the interfacing of a first and a second plurality of voice-band packets between the DSLAM and the plurality of standard telephone ports by:

converting the first digital bit stream received from one or more of the plurality of coders/decoders to a first plurality of voice-band packets and coupling the first plurality of voice-band packets to the LPN for transmission to the host digital terminal; and

converting the second plurality of voice-band packets received by the DSLAM to the second digital bit stream and coupling the second digital bit stream to one of the plurality of coders/decoders.

3. The system of claim 1, and further comprising a splitter/combiner at the wire center having a first port coupled to the subscriber loop link, a second port coupled to the DSLAM, and a third port coupled to one of the plurality of standard telephone ports of the wire center remote digital terminal.

4. The system of claim 1, and further comprising a splitter/combiner at a subscriber site having a first port coupled to the subscriber loop link, a second port coupled to the remote digital terminal, and a third port suitable for connection to a telephone device.

5. The system of claim 1, and further comprising a splitter/combiner at a subscriber site having a first port coupled to the subscriber loop link, a second port coupled to a DSL modem, and a third port suitable for connection to a telephone device.

6. The system of claim 1, wherein the wire center remote digital terminal comprises:

a plurality of subscriber line interface circuits, each of the plurality of subscriber line interface circuits suitable for connection to a telephone device;

a plurality of coder/decoders, each for connection to one of the plurality of subscriber line interface circuits, each of the plurality of coder/decoders capable of converting a first analog telephone signal received from the corresponding one of the plurality of subscriber line interface circuits to a first digital bit stream and converting a second digital bit stream to a second analog telephone signal to be supplied to one of the plurality of subscriber line interface circuits; and

a controller coupled to the plurality of coder/decoders and to the plurality of subscriber line interface circuits, the controller managing the interfacing of a first and a second plurality of voice-band packets between the DSLAM and the plurality of standard telephone ports by:

converting the first digital bit stream received from one or more of the plurality of coders/decoders to a first plurality of voice-band packets and coupling the first plurality of voice-band packets to the LPN for transmission to the host digital terminal; and

converting the second plurality of voice-band packets received by the DSLAM to the second digital bit stream and coupling the second digital bit stream to the corresponding one of the plurality of coders/decoders.

7. The system of claim 1, and further comprising a splitter/combiner at the wire center having a first port coupled to the subscriber loop link, a second port coupled to the DSLAM, and a third port coupled to one of the plurality of standard telephone ports of the wire center remote digital terminal.

8. The system of claim 1, and further comprising a splitter/combiner at a subscriber site having a first port coupled to the subscriber loop link, a second port coupled to the remote digital terminal, and a third port suitable for connection to a telephone device.

9. The system of claim 1, and further comprising a splitter/combiner at a subscriber site having a first port coupled to the subscriber loop link, a second port coupled to

a modem at the customer site, and a third port suitable for connection to a telephone device.

10. The system of claim 1, where the host digital terminal is within the LPN.

11. The system of claim 1, where the host digital terminal is connected to the LPN.

12. A system for communicating voice over a local packet network (LPN) to and from at least one customer site connected to the LPN via a local loop link, comprising:

a host digital terminal at a control site in communication with the LPN that interfaces a plurality of voice calls between the LPN and a public switched telephone network (PSTN) switch via the LPN; and

a wire center remote digital terminal that connects to or is incorporated in a digital subscriber line access multiplexer (DSLAM) that is in communication with the LPN, the wire center remote digital terminal providing a standard analog telephone service between each of a plurality of standard telephone ports and the PSTN switch through the host digital terminal, where each of the plurality of standard telephone ports is suitable for connection to a subscriber loop link, where the subscriber loop link is communicatively coupled with a subscriber site.

13. A method for communicating voice and data to and from a customer site over a local packet network (LPN) supporting a packet-based transport protocol, the customer site comprising a plurality of telephone devices, the method comprising steps of at the customer site:

converting a plurality of first analog telephone signals received from the plurality of telephone devices to a first plurality of voice-band packets and modulating the first plurality of voice-band packets for transmission over the LPN via a local loop link between the customer site and the LPN;

demodulating a second plurality of voice-band packets received from the LPN on the local loop link;

converting the second plurality of voice-band packets to a plurality of second analog telephone signals for connection to the corresponding one of the plurality of telephone devices; at a control site within or connected to the LPN:

receiving the first plurality of voice-band packets from the customer site via the LPN;

converting the first plurality of voice-band packets received from the customer site to a first timedivision multiplexed signal and coupling the first time-division multiplexed signal to a public switched telephone network (PSTN) switch; and

converting a second time-division multiplexed signal received from the PSTN switch destined for the customer site to a second plurality of voice-band packets and coupling the second plurality of voice-band packets to the LPN for transmission to the customer site; at a wire center:

providing a standard telephone port for connection to a subscriber loop link, where the subscriber loop link is communicatively coupled with a subscriber site, where the standard telephone port is capable of providing a standard analog service to the subscriber site;

demodulating a third plurality of voice-band packets transmitted from the control site via the LPN and received at a subscriber line port of a multiplexer, the multiplexer located at the wire center;

converting the third plurality of voice-band packets to a third analog telephone signal for connection to the subscriber loop link;

converting a fourth analog telephone signal received on the subscriber loop link to a fourth plurality of voice-band packets and modulating the fourth plurality of voice-band packets; and

coupling the fourth plurality of voice-band packets to the subscriber line port of the multiplexer for transmission over the LPN to the control site.

14. A method for communicating voice over a local packet network (LPN) to and from a customer site connected to the LPN via a local loop link, the customer site comprising a plurality of telephone devices, comprising steps of at a wire center:

connecting to a subscriber line port of a multiplexer located at the wire center;

providing a standard telephone port for connection to a subscriber loop link, where the subscriber loop link is communicatively coupled with a subscriber site, where the standard telephone port is capable of providing a standard analog telephone service to the subscriber site;

demodulating a first plurality of voice-band packets received at a subscriber line port of a multiplexer located at the wire center;

converting the first plurality of voice-band packets to a first analog telephone signal for connection to the subscriber loop link;

converting a second analog telephone signal received on the subscriber loop link to a second plurality of voice-band packets and modulating the second plurality of voice-band packets; and

coupling the second plurality of voice-band packets to the subscriber line port of the multiplexer for transmission over the LPN to a control site; at the control site within or connected to the LPN:

receiving the second plurality of voice-band packets from the wire center via the LPN;

converting the second plurality of voice-band packets received from the wire center to a first time-division multiplexed signal and coupling the first time-division multiplexed signal to a public switched telephone network (PSTN) switch; and

converting a second time-division multiplexed signal received from the PSTN switch destined for the wire center to the first plurality of voice-band packets and coupling the first plurality of voice-band packets to the LPN for transmission to the wire center.

15. An apparatus configured to transport a voice signal between at least one subscriber site and a host digital terminal via a local packet network (LPN), comprising:

an access network interface configured to communicate a plurality of voice packets to and from the host digital terminal via the LPN;

at least one telephone port configured to communicate an analog voice signal to and from a subscriber loop, where the subscriber loop is communicatively coupled with the at least one subscriber site; and

a converter configured to convert between the plurality of voice packets and the analog voice signal, where the remote digital terminal is communicatively coupled to the LPN by a local loop.

16. The apparatus of claim 15, where the local loop supports digital subscriber loop (DSL).

17. The apparatus of claim 15, where the remote digital terminal is located at a wire center.

18. The apparatus of claim 15, further comprising a DSL access multiplexer (DSLAM) communicatively coupled to the local loop and to the LPN.

19. The apparatus of claim 15, further comprising:

a subscriber line interface circuit (SLIC), the SLIC configured to communicate a first analog voice signal to a telephone device at the subscriber site, the SLIC further configured to communicate a second analog voice signal from the telephone device at the subscriber site;

a codec coupled to the SLIC, the codec configured to convert a first digital bit stream into the first analog voice signal, the codec further configured to convert the second analog voice signal into a second digital bit stream;

a controller coupled to the SLIC and to the codec, the controller configured to convert a first plurality of voice packets into the first digital bit stream, the controller further configured to convert the second digital bit stream into a second plurality of voice packets; and

a modem coupled to the controller and to the local loop link, the modem configured to modulate the first plurality of voice packets, the modem further configured to communicate the first plurality of voice packets to the LPN for transmission to the host digital terminal, the modem further configured to receive the second plurality of voice packets from the host digital terminal via the LPN, the modem further configured to demodulate the second plurality of voice packets.

20. The apparatus of claim 15, further comprising a plurality of telephone ports, each of the plurality of telephone ports configured to communicate one of a plurality of analog voice signals over a corresponding one of a plurality of subscriber loops, wherein the converter is further configured to convert between the plurality of voice packets and the plurality of analog voice signals, and wherein the remote digital terminal is communicatively coupled to each of a plurality of subscriber sites by one of the plurality of subscriber loops.

21. The apparatus of claim 20, further comprising:

a plurality of SLICs, each of the plurality of SLICs configured to communicate one of the plurality of first analog voice signals to a telephone device at the subscriber site, each of the plurality of SLICs further configured to communicate one of the plurality of second analog voice signals from the telephone device at the subscriber site; and

a plurality of codecs, each of the plurality of codecs coupled to a corresponding one of the plurality of SLICs, each of the plurality of codecs further configured to convert one of a plurality of first digital bit streams into one of the plurality of first analog voice signals, each of the plurality of codecs further configured to convert one of the plurality of second analog voice signals into a corresponding one of a plurality of second digital bit streams, wherein the controller is coupled to the plurality of SLICs and to the plurality of codecs, the controller configured to convert the first plurality of voice packets into a corresponding one of the first digital bit streams, the controller further configured to convert one of the plurality of second digital bit streams into the second plurality of voice packets.

22. A system to transport a voice signal between at least one subscriber site and a public switched telephone network (PSTN) over a local packet network (LPN) comprising:

a remote digital terminal, communicatively coupled to the LPN by a local loop and to the subscriber site by a subscriber loop; and

a host digital terminal, communicatively coupled to the PSTN and to the LPN, the remote digital terminal further comprising:

an access network interface configured to communicate a plurality of voice packets to and from the host digital terminal via the LPN;

at least one telephone port configured to communicate an analog voice signal to and from a subscriber loop, where the subscriber loop is communicatively coupled with the at least one subscriber site; and

a converter configured to convert between the plurality of voice packets and the analog voice signal; the host digital terminal further comprising:

a LPN interface coupled to the LPN, configured to communicate the plurality of voice-band packets to and from the remote digital terminal via the LPN;

a PSTN interface coupled to the PSTN, configured to communicate a voice signal to and from the PSTN; and

a packet converter coupled to the LPN interface and to the PSTN interface, the packet converter configured to convert the plurality of voice-band packets between a first format and a second format, where the first format is suitable for transport over the LPN and the second format is suitable for coupling to the PSTN.

23. The system of claim 22, where the host digital terminal is located in a central office.

24. The system of claim 22, where the first format is asynchronous transfer mode (ATM).

25. The system of claim 22, where the second format is time-division multiplex (TDM).

26. The system of claim 22, the remote digital terminal further comprising:

a subscriber line interface circuit (SLIC), the SLIC configured to communicate a first analog voice signal to a telephone device at the subscriber site, the SLIC further configured to communicate a second analog voice signal from the telephone device at the subscriber site;

a codec coupled to the SLIC, the codec configured to convert a first digital bit stream into the first analog voice signal, the codec further configured to convert the second analog voice signal into a second digital bit stream;

a controller coupled to the SLIC and to the codec, the controller configured to convert a first plurality of voice packets into the first digital bit stream, the controller further configured to convert the second digital bit stream into a second plurality of voice packets; and

a modem coupled to the controller and to the local loop link, the modem configured to modulate the first plurality of voice packets, the modem further configured to communicate the first plurality of voice packets to the LPN for transmission to the host digital terminal, the modem further configured to receive the second plurality of voice packets from the host digital terminal via the LPN, the modem further configured to demodulate the second plurality of voice packets.

27. The system of claim 22, the remote digital terminal further comprising a plurality of telephone ports, each of the plurality of telephone ports configured to communicate one of a plurality of analog voice signals over a corresponding one of a plurality of subscriber loops, wherein the converter is further configured to convert between the plurality of voice packets and the plurality of analog voice signals, and wherein the remote digital terminal is communicatively coupled to each of a plurality of subscriber sites by one of the plurality of subscriber loops.

28 The system of claim 27, the remote digital terminal further comprising:

a plurality of SLICs, each of the plurality of SLICs configured to communicate one of the plurality of first analog voice signals to a telephone device at the subscriber site, each of the plurality of SLICs further configured to communicate one of the plurality of second analog voice signals from the telephone device at the subscriber site; and

a plurality of codecs, each of the plurality of codecs coupled to a corresponding one of the plurality of SLICs, each of the plurality of codecs further configured to convert one of a plurality of first digital bit streams into one of the plurality of first analog voice signals, each of the plurality of codecs further configured to convert one of the plurality of second analog voice signals into a corresponding one of a plurality of second digital bit streams,

wherein the controller is coupled to the plurality of SLICs and to the plurality of codecs, the controller configured to convert the first plurality of voice packets into a corresponding one of the first digital bit streams, the controller further configured to convert one of the plurality of second digital bit streams into the second plurality of voice packets.

29. A system to transport a voice signal between at least one subscriber site and a public switched telephone network (PSTN) over a local packet network (LPN), and to transport a data signal between at least one subscriber site and a data network, comprising:

a remote digital terminal, communicatively coupled to the LPN by a local loop and to the subscriber site by a subscriber loop; and

a host digital terminal, communicatively coupled to the PSTN and to the LPN; and

an access multiplexer, communicatively coupled to the remote digital terminal, the remote digital terminal further comprising:

a network interface, configured to communicate a plurality of data packets between a local area network at the subscriber site and the data network via the LPN;

an access network interface configured to communicate a plurality of voice packets to and from the host digital terminal via the LPN;

at least one telephone port configured to communicate an analog voice signal to and from a subscriber loop, where the subscriber loop is communicatively coupled with the at least one subscriber site; and

a converter configured to convert between the plurality of voice packets and the analog voice signal;

the host digital terminal further comprising:

a LPN interface coupled to the LPN, configured to communicate the plurality of voice-band packets to and from the remote digital terminal via the LPN;

a PSTN interface coupled to the PSTN, configured to communicate a voice signal to and from the PSTN; and

a packet converter coupled to the LPN interface and to the PSTN interface, the packet converter configured to convert the plurality of voice-band packets between a first format and a second format, where the first format is suitable for transport over the LPN and the second format is suitable for coupling to the PSTN.

30. The system of claim 29, further comprising:

a first splitter, having a first port coupled to the local loop, a second port coupled to the access multiplexer, and a third port coupled to the at least one telephone port; and

a second splitter, having a fourth port coupled to the local loop, a fifth port coupled to a telephone device, and a sixth port coupled to the remote digital terminal.

31. The system of claim 29, further comprising:

a first splitter, having a first port coupled to the local loop, a second port coupled to the access multiplexer, and a third port coupled to the at least one telephone port; and

a second splitter, having a fourth port coupled to the local loop, a fifth port coupled to a telephone device, and a sixth port coupled to a digital subscriber loop (DSL) modem, the DSL modem located at the customer site.

32. The system of claim 29, the remote digital terminal further comprising:

a subscriber line interface circuit (SLIC), the SLIC configured to communicate a first analog voice signal to a telephone device at the subscriber site, the SLIC further configured to communicate a second analog voice signal from the telephone device at the subscriber site;

a codec coupled to the SLIC, the codec configured to convert a first digital bit stream into the first analog voice signal, the codec further configured to convert the second analog voice signal into a second digital bit stream;

a controller coupled to the SLIC and to the codec, the controller configured to convert a first plurality of voice packets into the first digital bit stream, the controller further configured to convert the second digital bit stream into a second plurality of voice packets; and

a modem coupled to the controller and to the local loop link, the modem configured to modulate the first plurality of voice packets, the modem further configured to communicate the first plurality of voice packets to the LPN for transmission to the host digital terminal, the modem further configured to receive the second plurality of voice packets from the host digital terminal via the LPN, the modem further configured to demodulate the second plurality of voice packets.

33. The system of claim 29, the remote digital terminal further comprising:

a plurality of telephone ports, each of the plurality of telephone ports configured to communicate one of a plurality of analog voice signals over a corresponding one of a plurality of subscriber loops, wherein the converter is further configured to convert between the plurality of voice packets and the plurality of analog voice signals, and wherein the remote digital terminal is communicatively coupled to each of a plurality of subscriber sites by one of the plurality of subscriber loops.

34. The system of claim 29, the remote digital terminal further comprising:

a plurality of SLICs, each of the plurality of SLICs configured to communicate one of the plurality of first analog voice signals to a telephone device at the subscriber site, each of the plurality of SLICs further configured to communicate one of the plurality of second analog voice signals from the telephone device at the subscriber site; and

a plurality of codecs, each of the plurality of codecs coupled to a corresponding one of the plurality of SLICs, each of the plurality of codecs further configured to convert one of a plurality of first digital bit streams into one of the plurality of first analog voice signals, each of the plurality of codecs further configured to convert one of the plurality of second analog voice signals into a corresponding one of a plurality of second digital bit streams,

wherein the controller is coupled to the plurality of SLICs and to the plurality of codecs, the controller configured to convert the first plurality of voice packets into a corresponding one of the first digital bit streams, the controller further configured to convert one of the plurality of second digital bit streams into the second plurality of voice packets.

35. A method for communicating voice over a local packet network (LPN) to and from at least one subscriber site, comprising:

at a wire center:

receiving at least one first analog telephone signal from at least one subscriber site over at least one subscriber loop;

converting the at least first analog telephone signal to a first plurality of voice-band packets;

communicating the first plurality of voice-band packets to the LPN for transmission to a control site;

receiving a second plurality of voice-band packets from the control site via the LPN;

converting the second plurality of voice-band packets to at least one second analog telephone signal; and

communicating the at least one second analog telephone signal to the at least one subscriber site via the at least one subscriber loop;

at a control site:

receiving the first plurality of voice-band packets from the wire center via the LPN;

converting the first plurality of voice-band packets to a first TDM signal;

communicating the first TDM signal to a PSTN switch;

receiving a second TDM signal from the PSTN switch;

converting the second TDM signal into the second plurality of voice-band packets; and

communicating the second plurality of voice-band packets to the LPN for transmission to the wire center.

36. The method of claim 35 further comprising:

modulating the first plurality of voice-band packets; and

demodulating the second plurality of voice-band packets.

37. The method of claim 35 where communicatively coupling the first plurality of voice-band packets to the LPN further comprises communicating the first plurality of voice-band packets to a subscriber line port of an access multiplexer, where the access multiplexer is in communication with the LPN.

38. A method for transporting voice signals over a local packet network (LPN) to and from a subscriber site, the subscriber site comprising a plurality of first telephone

devices and a second telephone device, the method comprising:

at a customer site:

receiving a first plurality of analog telephone signals from the plurality of first telephone devices over a subscriber loop;

converting the first plurality of analog telephone signals to a first plurality of voice-band packets;

coupling the first plurality of voice-band packets to the LPN for transmission to a control site;

receiving a second plurality of voice-band packets from the control site via the LPN;

converting the second plurality of voice-band packets to a second plurality of analog telephone signals; and

communicating the second plurality of analog telephone signals to the plurality of first telephone devices via the subscriber loop;

at a wire center:

receiving a first analog telephone signal from the second telephone device over the subscriber loop;

converting the first analog telephone signal to a third plurality of voice-band packets;

communicating the third plurality of voice-band packets to the LPN for transmission to the control site;

receiving a fourth plurality of voice-band packets from the control site via the LPN;

converting the fourth plurality of voice-band packets to a second analog telephone signal; and

communicating the second analog telephone signal to the second telephone device via the subscriber loop;

at a control site:

receiving the first plurality of voice-band packets from the customer site via the LPN;

receiving the third plurality of voice-band packets from the wire center via the LPN;

converting the first plurality of voice-band packets to a first TDM signal;

converting the third plurality of voice-band packets to a second TDM signal;

communicating the first TDM signal and the second TDM signal to a PSTN switch;

receiving a third TDM signal and a fourth TDM signal from a PSTN switch;

converting the third TDM signal into the second plurality of voice-band packets;

converting the fourth TDM signal into the fourth plurality of voice-band packets;

communicating the second plurality of voice-band packets to the LPN for transmission to the customer site; and

communicating the fourth plurality of voice-band packets to the LPN for transmission to the wire center.

* * * * *

EXHIBIT A
PAGE 23 OF 24

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 6,639,913 B1 Page 1 of 1
DATED : October 28, 2003
INVENTOR(S) : David P. Frankel, Joe Boucher and Kenneth M. Kolderup

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 2,
Line 66, delete the comma "," between the words "the" and "L.P.N. --

Column 7,
Line 26, delete "lo" between the words "is" and "the".

Column 10,
Line 62, delete the period "." between the words "each" and "call".

Column 14,
Line 66, delete the period "." between the words "a" and "data".

Signed and Sealed this

Twenty-third Day of March, 2004

Jon W. Dudas

JON W. DUDAS
Acting Director of the United States Patent and Trademark Office

EXHIBIT A
PAGE 24 OF 24

US006580785B2

(12) **United States Patent**
Bremer et al.

(10) **Patent No.: US 6,580,785 B2**
(45) **Date of Patent: Jun. 17, 2003**

(54) **APPARATUS AND METHOD FOR SIMULTANEOUS MULTIPLE TELEPHONE TYPE SERVICES ON A SINGLE TELEPHONE LINE**

(75) Inventors: **Gordon Bremer**, Clearwater, FL (US); **Jeff Davis**, Indian Rocks Beach, FL (US)

(73) Assignee: **Paradyne Corporation**, Largo, FL (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 57 days.

(21) Appl. No.: **09/032,671**

(22) Filed: **Feb. 27, 1998**

(65) **Prior Publication Data**

US 2001/0022836 A1 Sep. 20, 2001

Related U.S. Application Data

(60) Provisional application No. 60/039,265, filed on Feb. 28, 1997.

(51) **Int. Cl.**^{7} **H04M 1/64**; H04M 11/00

(52) **U.S. Cl.** **379/88.13**; 379/49; 379/87; 379/88.07; 379/88.08; 379/93.04; 379/93.09; 379/100.15; 379/102.03

(58) **Field of Search** 379/29, 33, 34, 379/37, 47, 49, 74, 88.13, 88.17, 92.01, 92.03, 92.02, 93.08, 93.09, 93.11, 100.15, 100.17, 102.01, 102.03, 87, 88.07, 88.08, 93.04; 370/271, 286, 276, 426, 351; 348/15, 20

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Primary Examiner—Allan Hoosain
(74) *Attorney, Agent, or Firm*—Thomas, Kayden, Horstemeyer & Risley, LLP

(57) **ABSTRACT**

Apparatus and method for simultaneously providing multiple telephone-type services to any/all POTS-type devices on a single wire pair at a user premises. The present invention provides for the ability to add separately addressable POTS devices on a single service loop. This can be accomplished in at least two ways: first by the use of a multipoint protocol or second by Frequency Division Multiplexing.

93 Claims, 8 Drawing Sheets

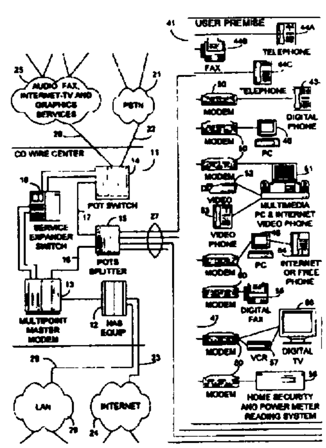

EXHIBIT B
PAGE 1 OF 20

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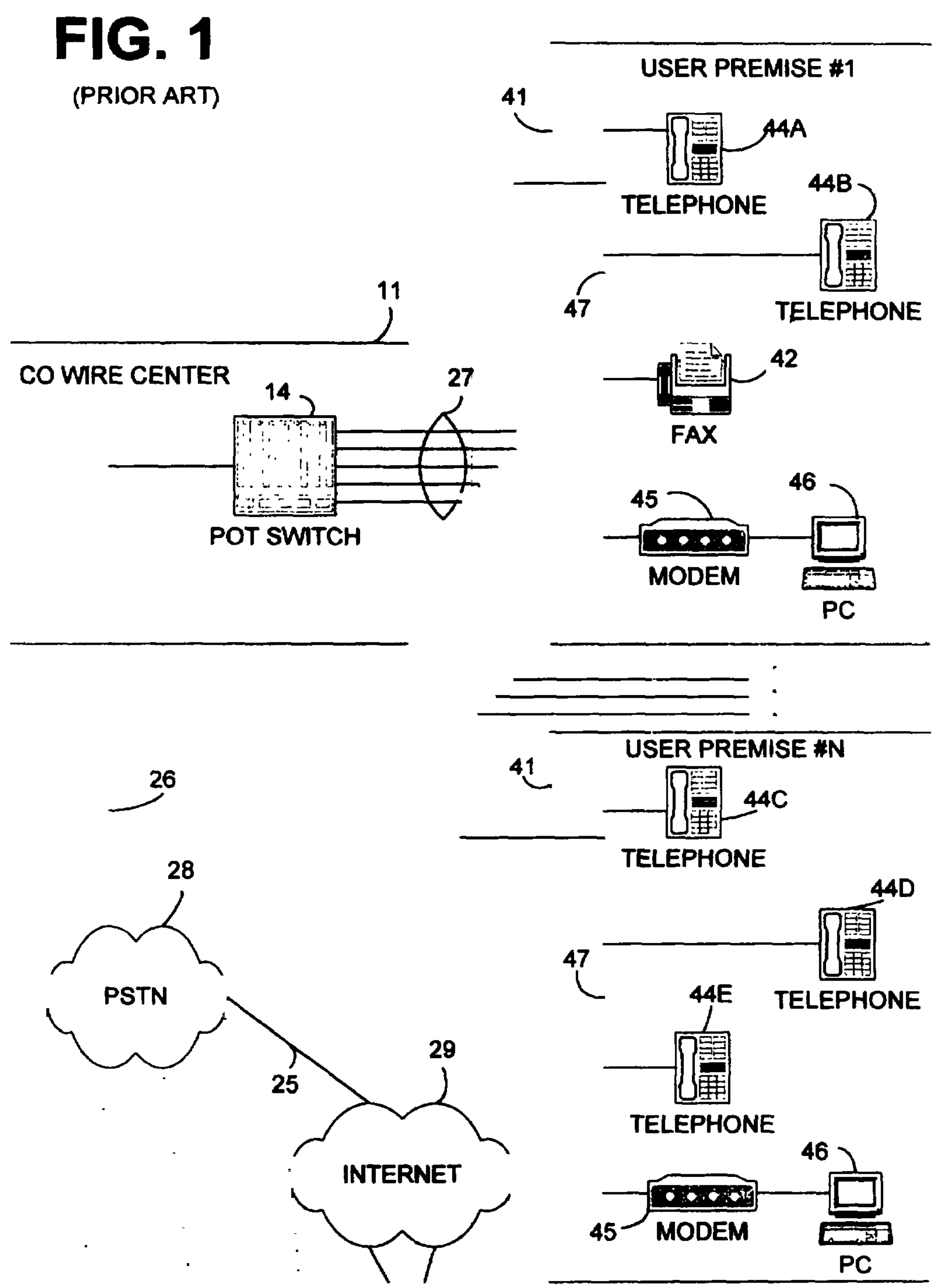
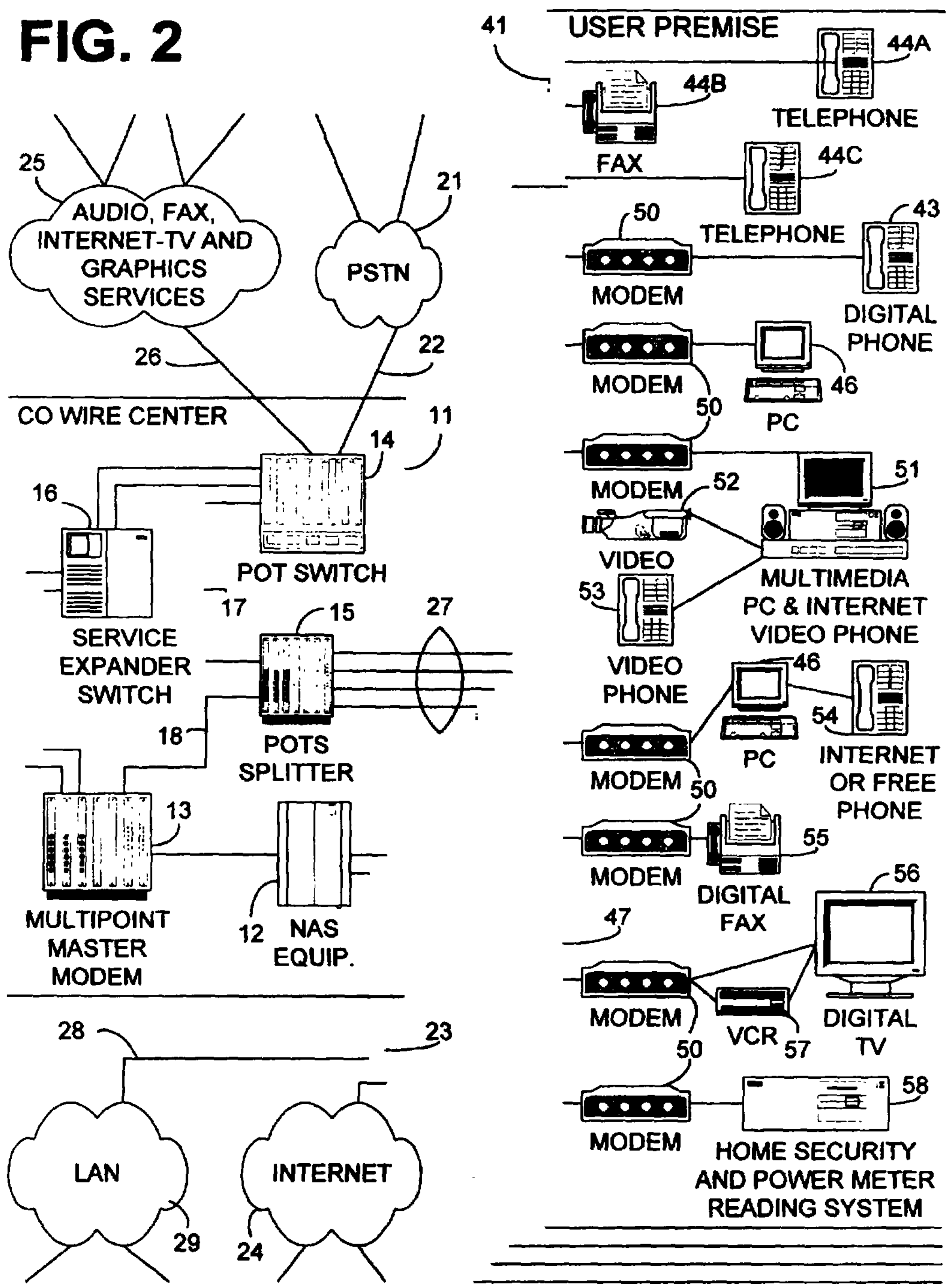


FIG. 1
(PRIOR ART)
USER PREMISE #1
41
44A
TELEPHONE
44B
TELEPHONE
47
11
CO WIRE CENTER
14
27
42
FAX
POT SWITCH
45
46
MODEM
PC
USER PREMISE #N
26
41
44C
TELEPHONE
28
44D
47
PSTN
44E
TELEPHONE
29
25
TELEPHONE
46
INTERNET
45
MODEM
PC

FIG. 2


41
USER PREMISE
44A
TELEPHONE
44B
FAX
44C
TELEPHONE
43
DIGITAL PHONE
50
MODEM
46
PC
51
52
VIDEO
MULTIMEDIA PC & INTERNET VIDEO PHONE
53
VIDEO PHONE
54
INTERNET OR FREE PHONE
55
DIGITAL FAX
56
47
VCR
57
DIGITAL TV
58
HOME SECURITY AND POWER METER READING SYSTEM
25
AUDIO, FAX, INTERNET-TV AND GRAPHICS SERVICES
21
PSTN
26
22
CO WIRE CENTER
11
14
16
POT SWITCH
17
15
27
SERVICE EXPANDER SWITCH
18
POTS SPLITTER
13
MULTIPOINT MASTER MODEM
12
NAS EQUIP.
28
23
LAN
INTERNET
29
24

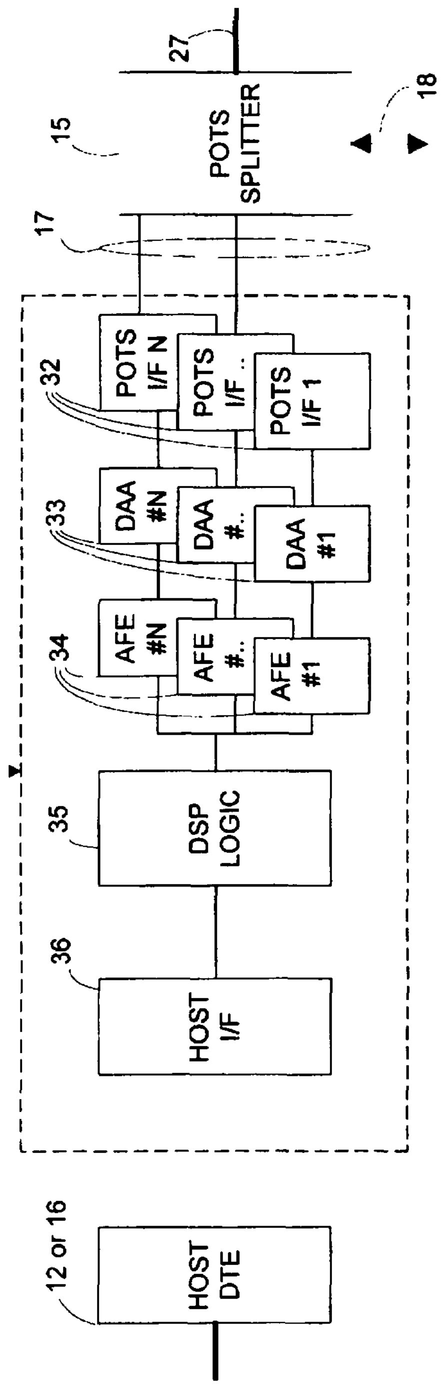

13
HOST DTE
12 or 16
HOST I/F
36
DSP LOGIC
35
AFE #1
AFE #..
AFE #N
34
DAA #1
DAA #..
DAA #N
33
POTS I/F 1
POTS I/F ..
POTS I/F N
32
17
POTS SPLITTER
15
18
27

FIG. 3A

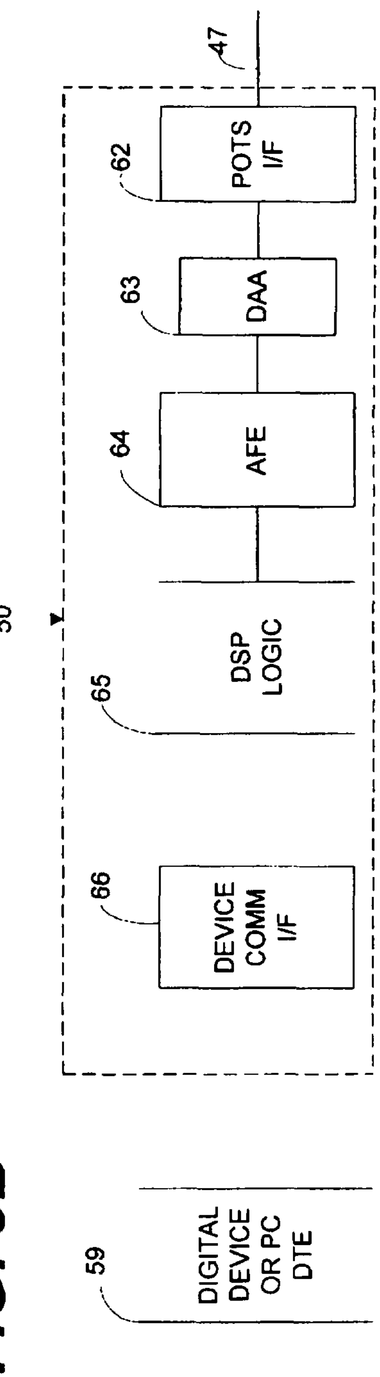

50
DIGITAL DEVICE OR PC DTE
59
DEVICE COMM I/F
66
DSP LOGIC
65
AFE
64
DAA
63
POTS I/F
62
47

FIG. 3B

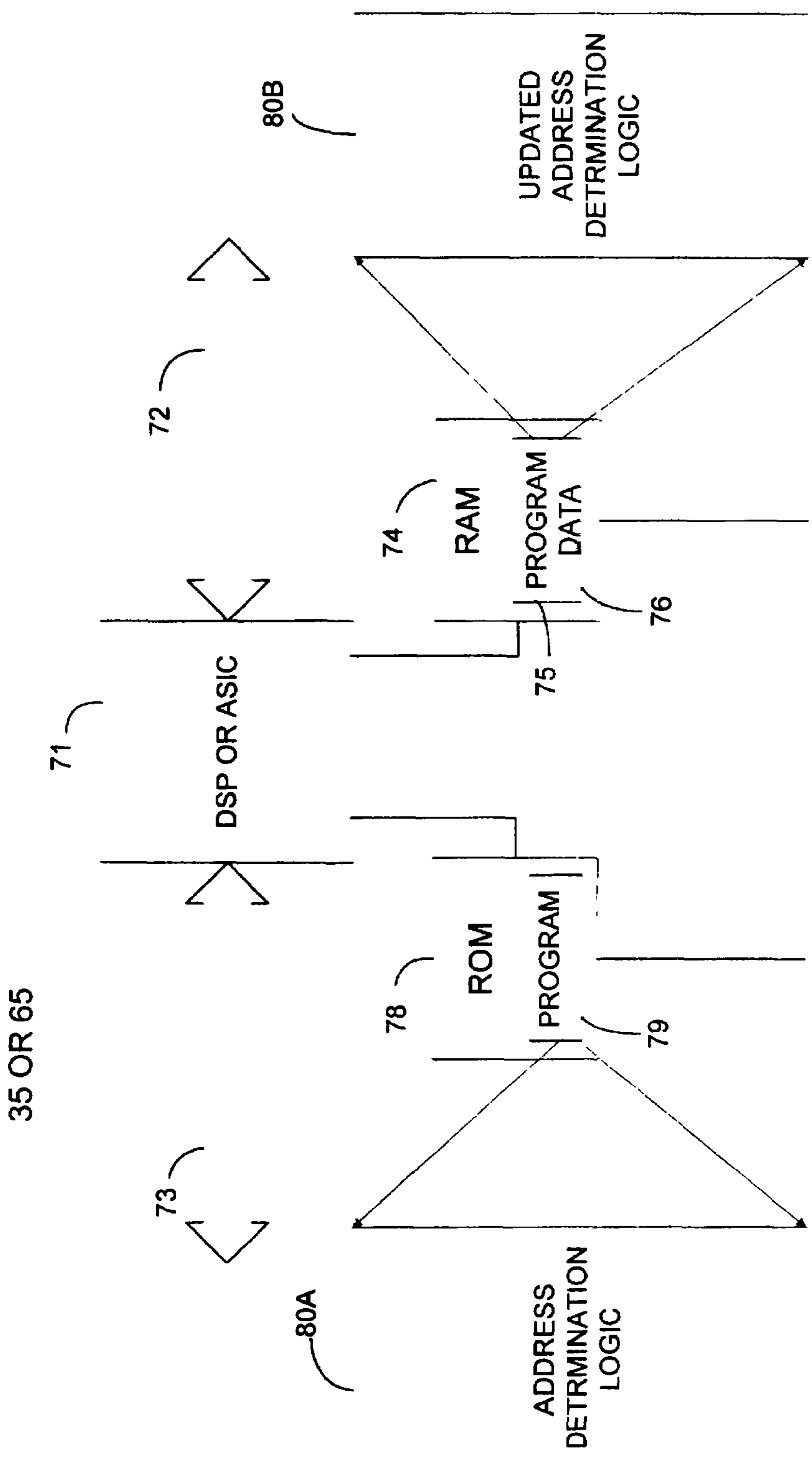

35 OR 65
71
DSP OR ASIC
72
73
74
RAM
PROGRAM
DATA
75
76
78
ROM
PROGRAM
79
80A
ADDRESS
DETRMINATION
LOGIC
80B
UPDATED
ADDRESS
DETRMINATION
LOGIC

FIG. 4

FIG. 5

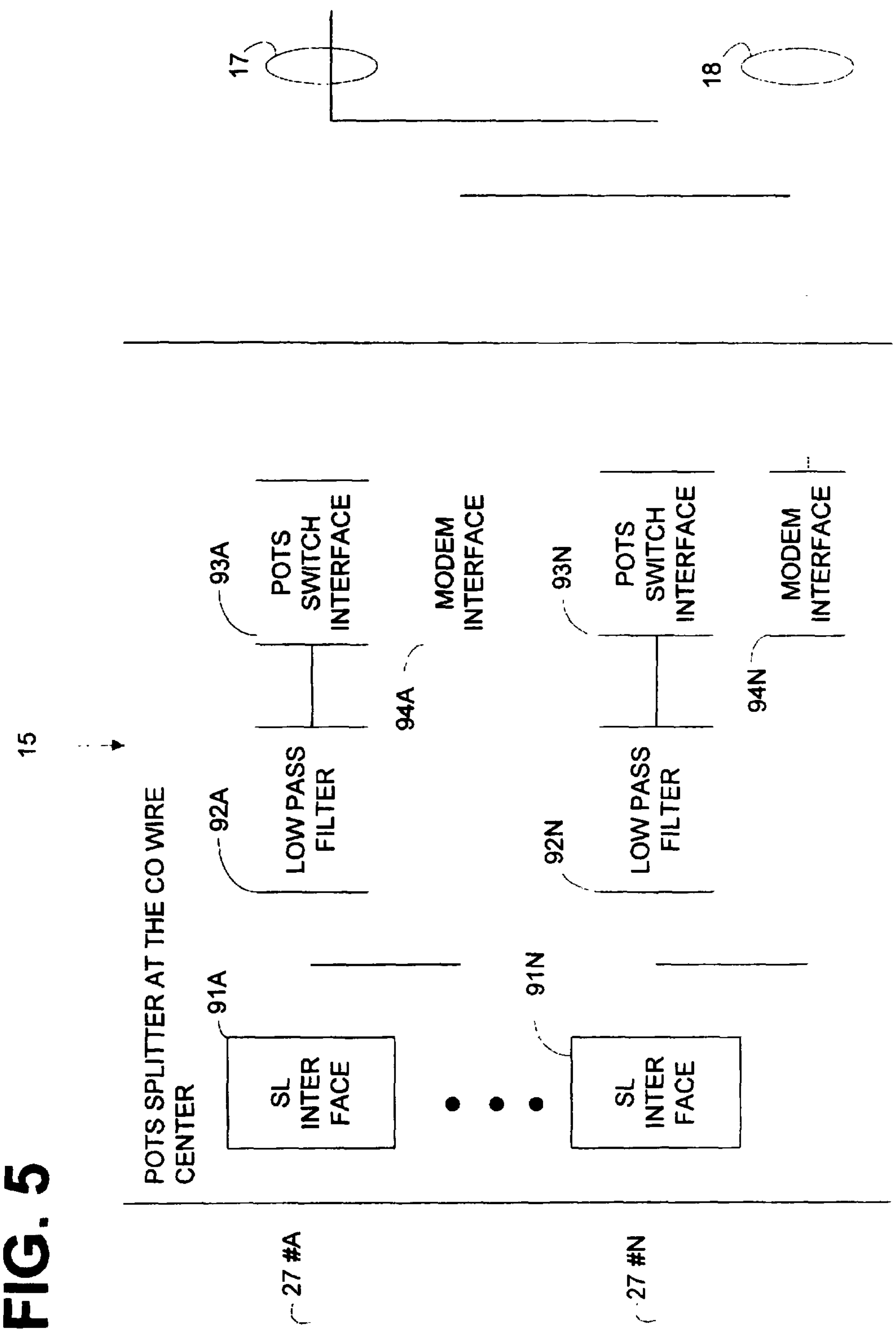
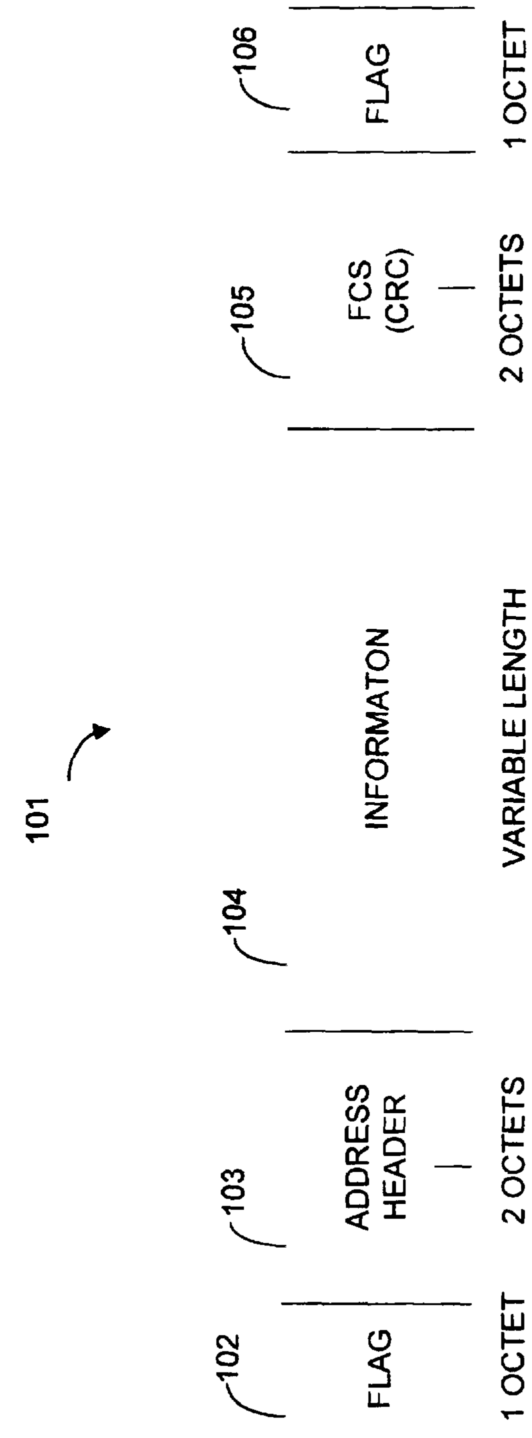

15
POTS SPLITTER AT THE CO WIRE CENTER
27 #A
27 #N
91A
91N
SL INTER FACE
SL INTER FACE
92A
92N
LOW PASS FILTER
LOW PASS FILTER
93A
93N
POTS SWITCH INTERFACE
POTS SWITCH INTERFACE
94A
94N
MODEM INTERFACE
MODEM INTERFACE
17
18

Fig. 6


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

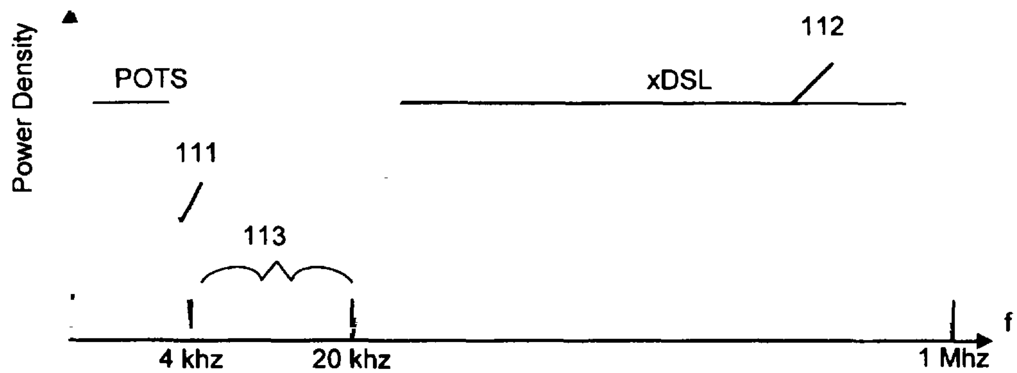


FIG. 7

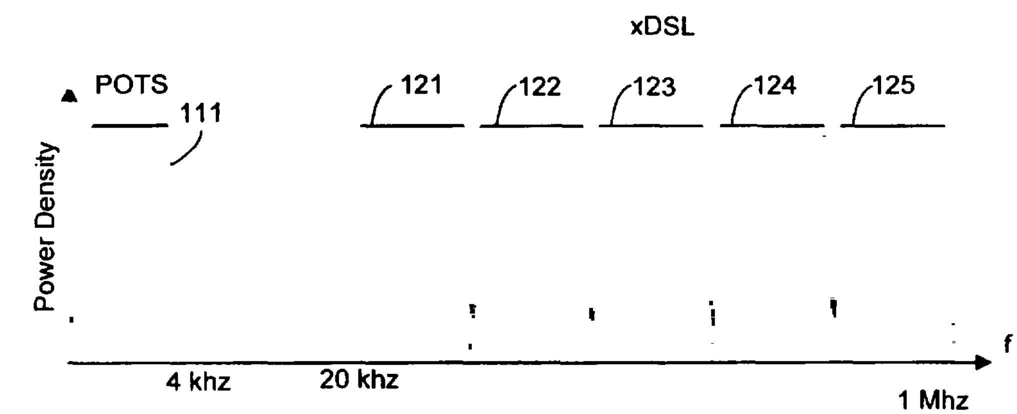


FIG. 8

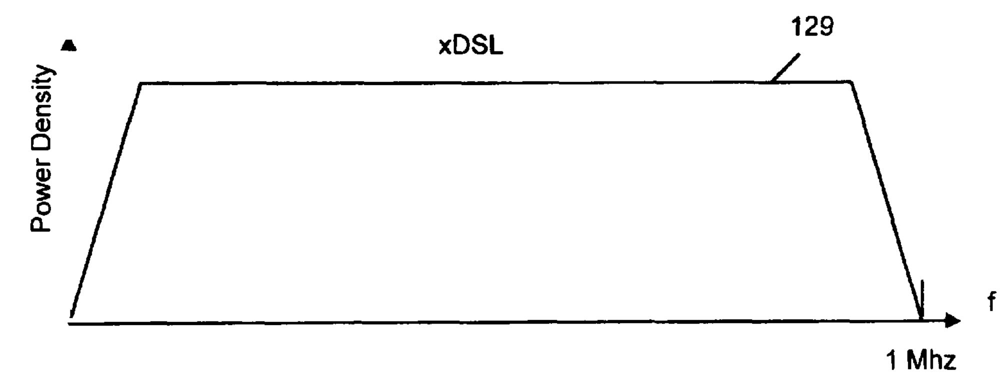


FIG. 9

FIG. 10

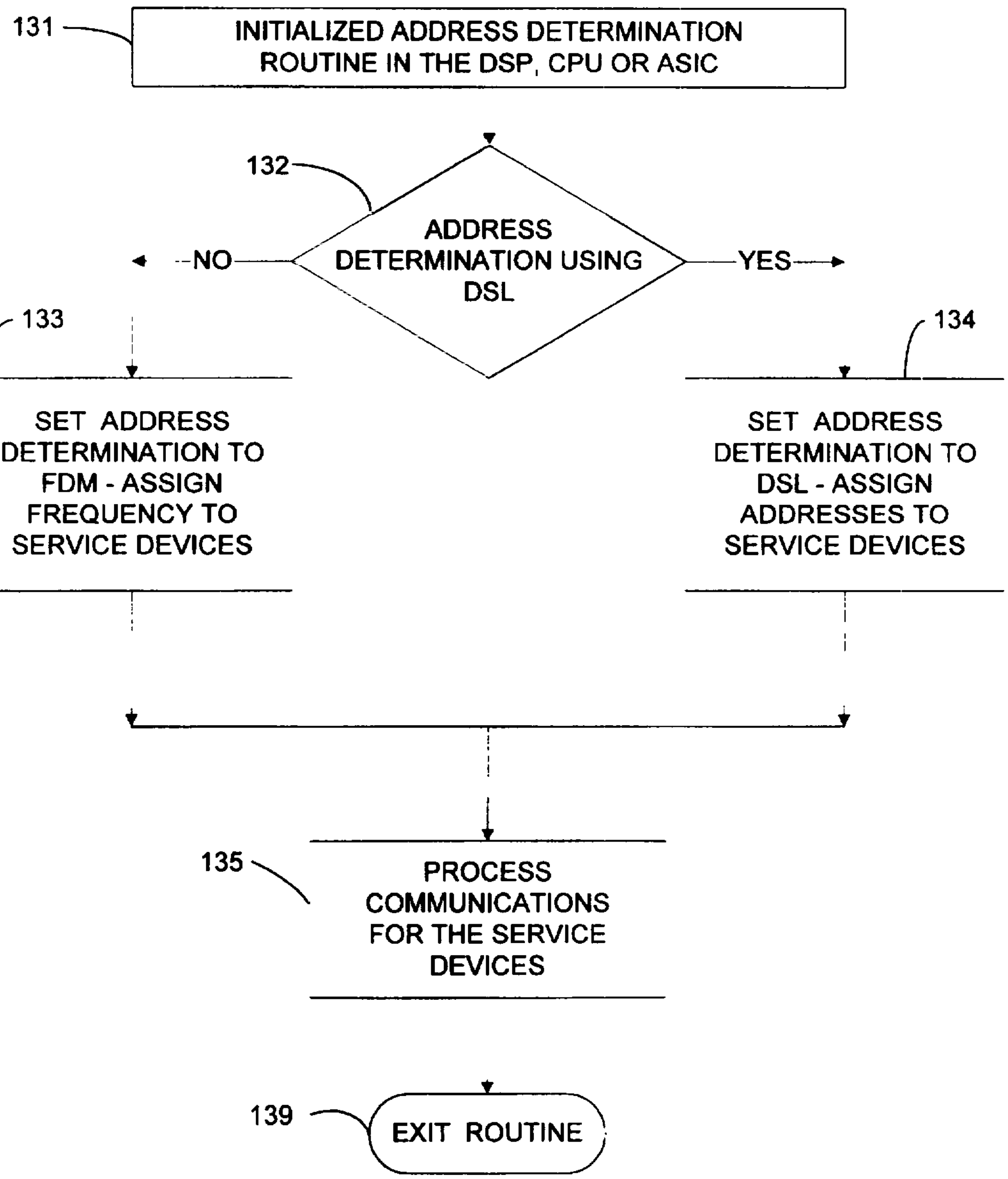


EXHIBIT B
PAGE 10 OF 20

APPARATUS AND METHOD FOR SIMULTANEOUS MULTIPLE TELEPHONE TYPE SERVICES ON A SINGLE TELEPHONE LINE

CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of U.S. Provisional Patent Application Serial No. 60/039,265, filed on Feb. 28, 1997, and entitled "SIMULTANEOUS MULTIPLE TELEPHONE TYPE SERVICES ON A SINGLE TELEPHONE LINE".

BACKGROUND OF THE INVENTION

1 Field of the Invention

The present invention generally relates to an apparatus and method for enabling a plurality of analog and digital sets of services that can be utilized simultaneously on a single telephone line.

2. Description of the Related Art

Presently, telephone companies can offer only one set of analog services to any and all POTS-type devices on each subscriber line wire pair at the premise, because current POTS service requires one (1) line per POTS service set. This is because device types are mutually exclusive, and consequently only one device type can utilize the service line at any one time (i.e. one active telephone, or a single fax operation at a time). A further limitation exists for the telephones, such that all extensions are connected to the same conversation. Presently if multiple sets of services are desired, an additional line is required for each additional set of services. This is most evident in situations like 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 sets of service to the premise on a single line. Accordingly, there is a need to develop an apparatus and method to transmit a plurality of data signals in parallel with the analog POTS signal, thereby providing multiple telephone-type sets of services on a single telephone line.

With such an apparatus and method for enabling simultaneous multiple sets of telephone-type services on a single telephone line, the telephone companies can offer numerous sets of services to any/all devices on each wire pair at the premise.

SUMMARY OF THE INVENTION

Certain objects, advantages and novel features of the invention will be set forth in part in the description that follows and in part will become apparent to those skilled in the art upon examination of the following or may be learned with the practice of the invention. The objects and advantages of the invention may be realized and obtained by means of the instrumentality's and combinations particularly pointed out in the appended claims.

To achieve the advantages and novel features, the present invention is generally directed to a data communications apparatus and method that allows a user to utilize simultaneously multiple telephone-type services to any/all POTS-type devices on each wire pair at the premise. The present invention provides for the ability to add separately addressable POTS devices on a single service loop. This can be accomplished in at least two ways: first by the use of a multipoint protocol or second by Frequency Division Multiplexing.

One embodiment of the present invention accomplishes this by using a multipoint protocol and providing each premises device with a unique device ID that is separately addressable.

Another embodiment of the present invention accomplishes this by using the frequency division multiplexing (FDM) method, that utilizes a device that assigns an available frequency range, within the bandwidth of the communication medium, for each device that is separately addressable.

Another embodiment of the present invention accomplishes this by using the time division multiplexing (TDM) method, that combines separate signals (i.e. analog and digital) into a single high-speed data transmission in which the transmission time is broken into segments. Each segment carries one element of one signal. The separate signals are sampled in order at regular intervals that are then combined in the single high-speed single. Each time period is then assigned for each device that is separately addressable. The above TDM technique does not provide simultaneous access via connection to phone jacks. The modem apparatus used in this embodiment includes a memory containing a plurality of program routine sequences and a processor that performs the selected program routine sequences to enable the simultaneous multiple access techniques disclosed by 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", Ser. No. 08/962,796, filed on Nov. 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", Ser. No. 09/036,226 filed on, Feb 26, 1998, 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 CO POTS and digital signals splitter of FIG. 2.

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 block diagram of the frequency spectrums utilized by the multipoint protocol packets of FIG. 6.

FIG. 8 is a block diagram of the frequency spectrums utilized by the frequency division multiplexing method that provides each device with a separately addressable access.

FIG. 9 is a block diagram of the Frequency spectrum utilized by the multipoint protocol packets of FIG. 5, when not currently utilizing POTS devices.

FIG. 10 is a flow chart of the process for the initializing the address determination routines residing in the DSP, CPU or ASIC device of FIG. 4.

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 dial 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) 43, 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 previously discussed.

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 14 that routes all POTS signals, including both those to/from analog devices such as telephones and to/from digital data devices such as dial modems or fax machines. The POTS signals are sent from the POTS switch 14 to the other central office wire centers and to remote premises and to data services such as the Internet services via the public switch telephone network (PSTN) 28. The CO wire center thus can offer only a single telephone number and only one set of services for each subscriber line 27.

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

When a user wishes to place a telephone call on device 44, 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 call request by allowing a flow of D.C. current and a dial tone of 480 Hz to be sent to device 44. The outgoing telephone call signals are transmitted, as described before, via subscriber line 27 to POTS switch 14. The analog POTS system signals are transmitted, via the PSTN 28, to the destination central office wire center 11 of the destination user premises 41. The signal is further directed towards a POTS switch 14 within the destination central office wire center 11. The signal is transmitted, via subscriber line 27, to 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 communicate data over a digital network via his personal PC 46 or the like, the dial modem 45 puts 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 44. Digital signals from the digital device are transformed into analog signals by dial modem 45. The signals are transmitted via the PSTN 28 to destination user premises 41 in the same manner as the analog signals in the aforementioned example. The signals may alternatively be routed to the Internet 29 via an Internet Service provider to provide access to Internet data.

FIG. 2 illustrates the plain old telephone system (POTS) networks including data communication DSL modems 13 and 50 of the preferred embodiment. The data communication DSL modems 50 include the 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 digital telephone services, Internet television, audio and multimedia, fax, graphic services, high-speed Internet services, high-speed land services, Internet telephone service, stereo/audio service, power meter reading, home management and security services. Again, the operation of such services are generally understood and are further not necessary in order to describe the operation of the present invention. As further illustrated in FIG. 2, the prior POTS voice devices of the prior art 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 as needed to provide the simultaneous multiple access. Other methods of providing multiple access, such as frequency division multiplexing or other multiplexing techniques, may be utilized with some limitation in overall performance. The different equipment devices at the user premises can be identified and accessed by a multiple access code (MAC) address as determined by the DSL 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.

For audio services, the modem 50 can be coupled with audio compression for a telephone or stereo receiver as shown by device 51.

Digital phone 43 utilizes modem 50 to digitize an audio buffer as necessary and transmits the digitized audio at an average data rate of 8 KPS and performs a reverse function in the received direction. Thus, the digital phone acts to the user as a telephone with digital clarity and services provided. The digital phone may communicate over PSTN via compatible analog digital conversions in the optional Teleco switch expander 16.

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

Multimedia and video telephone service can be provided utilizing video camera 52 to capture video, the video tele-

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phone 53 which may be a microphone and multimedia PC Internet video phone device 51, which captures video and audio and provides the digitized information to modem 50 for transmission to the destination user. The Internet video phone may use either the PSTN or Internet or other land-type network for data communications. Internet phone 54 has the features of the digital phone with a protocol required for communication over Internet or land networks.

Digital faxes can be transmitted and received via the digital fax device 55 through modem 50 which would digitize the information and transmit it via the Internet land or PSTN networks.

Digital television 56 and digital video cassette recorder (VCR) 57 can be utilized with the Internet streaming to receive and record Internet television and audio/visual data streaming. Services that require low-delay and medium delay (latency) utilize the "quality of service" polling techniques to assure that real-time applications are serviced in a timely manner. The "quality of service" polling techniques are disclosed by the modem described in commonly assigned and co-pending U.S. Patent Application entitled "APPARATUS AND METHOD FOR DSP SHARING USING STATISTICAL PROPERTIES OF DATA", Ser. No. 09/027,705 filed on Feb. 23, 1998, herein incorporated by reference.

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 as shown and described with regard to FIG. 2, are accessed via unique addresses. For each particular telephone company service provided, that service provides the user a unique address or frequency range 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 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 15 which separates the POTS communications that are now transmitted in the frequency band between 1 kHz and 4 kHz. These POTS signals are identified in POTS splitter 15 and separated from the multiple service signals operating at a higher frequency at POTS splitter 15. The POTS voice signals are separated from the data signals and transmitted to POTS switch 14 for communications over the PSTN or WEB TV, audio, fax, graphic services, home security and power meter reading networks 25. The LAN data signals and Internet data signals are separated from the voice POTS signals in POTS splitter 15 and forwarded on the master point modem 13 for further transmission through the NAS equipment devices to the Internet 24 and other LAN networks 29.

Service signals from the digital phone multimedia Web TV, digital fax, home security and power meter reading systems are provided to the multipoint master modem 13 by the POTS splitter 15. These signals are forwarded on to the service expander switch 16 for further transmission through the POTS switch 14 on communication link 26 to the Web TV, audio, fax, graphic services, digital TV, Internet phone and the like network 25. The digital phone and Internet and free phone each may have a standard telephone number or may share a number with the other devices. The digital phone and Internet or free phone would have standard Teleco POTS features and billing. The free phone 54 would have a different multiple access code and would permit free long distance calls on the Internet 24.

FIG. 3A is a block diagram of the CO wire center multichannel data communications device modem (modem 13) constructed in accordance with the present invention. The typical configuration of the central wire office 11 multichannel data communication device 13 is connected, via a POTS splitter 15, to the subscriber line 27. The analog signals output from POTS splitter 15 into the central office multichannel data communications device 13, are connected through communication links into the POTS interface 32. The central office multichannel data communications device 13 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 13. Because multiple analog input lines are permitted, device 13 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. Line protection circuit (not shown) protects the multichannel communications device 13 against line surges, lightning strikes, and the like. Line protection circuit (not shown) is then further connected to the impedance and isolation circuit (not shown), via a communication link. The impedance and isolation circuit (not shown) also contains circuitry (not shown) to detect ring indicator on 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 hyped interface (not shown). The dial access arrangement (DAA) logic 33 connects the analog signals to the 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. 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 (not shown) and driver digital-to-analog converter (not shown) are both connected to the bi-directional digital communication link. Ring indicator and off-hook conditions are processed in ring indicator (RI) off-hook (OH) impedance controller (not shown).

The AFE logic 34 transmits the digital signal to the DSP logic 25 for reconstruction of the digital data. Multiple analog front ends logic 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 33 and analog front ends logic 34 are not necessary to practice the present invention, and it 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.

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 12 or 16 devices for further transmission over the PSTN 21, Internet 24, LAN 29 or other services network 25.

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 only one POTS line connection.

In the typical configuration, the user premises line 47 is connected to line jack POTS interface 62. The line jack POTS 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 item 3X. The digital signal processor logic 65 is connected to the host by a local IF bus via a communication line, through the data terminal equipment (DTE) interface 66, which connects to a device such as a fax, digital phone, personal computer (PC), or the like.

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 50. Central office location 11 is typically the local telephone company's 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) 78 and random access memory (RAM) 74. ROM 78 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.

In the first embodiment, the address determination routines 80A are in the digital signal processor engine 65 program ROM 79. Address determination 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 75 program area 80B. It is in this way that an updated routine may be downloaded to the modem apparatus to update the address determination routines.

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 through across digital communication link 73. Digital communication links 72 and 73 can be comprised of 8, 16, 32, 64, 128 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 block diagram of the POTS splitter 15 at the central office wire center 11. The POTS splitter has numerous subscriber line interfaces 91I–91N that are connected to subscriber lines 27I–27N. The POTS splitter 15 accepts analog signals across subscriber line 27I–27N, conducts the analog signal through low pass filter 92 for transmission to the POTS switch interface 93. The POTS switch then transmits analog signals across communication link 17 to the POTS switch 14. The analog signals received from subscriber line interface 71 are also transmitted through modem interface 94, which transmits the data communication traffic, via communication link 18, to the master modem 13.

With reference now to FIG. 6, shown is a schematic view illustrating a communications packet 101 transported by the modem 50 of FIG. 3. Packet 101 is a standard frame relay communication packet. Begin flag 102 signals the start of the packet. Frame 103 is the address header and is depicted as two octets. An octet is an eight bit word. Frame 103 can be a length of two to four octets, however, for simplicity is shown as two octets in this preferred embodiment. Following frame 103 is information frame 104 which contains the user data to be transported over the network, and any proprietary header information required. Information frame 104 is variable in length depending upon the information to be transported. Following information frame 104 is frame check sequence (FCS) frame 105. The FCS frame is typically two octets in length and is typically 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.

Turning now to the drawings, FIG. 7 is a diagram illustrating frequency band communications. The terminology "frequency band communications" is used here to indicate communications of information within a certain defined, frequency band. As is known in the prior art, POTS communications are transmitted in the frequency band 111 defined between about 0 Hz (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 band 113 is required 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 about 1 MHz. As a result, and for the reasons described above, without the addition of extra equipment, such as POTS filters, splitters, etc. The xDSL signals are not compatible with attached POTS-type equipment, such as telephones, PSTN modems, facsimile machines, etc

As will be discussed in more detail below, alternative embodiment of the present invention provides an upper transmission band having an upper frequency boundary that is much lower than the 1 MHz frequency boundary often encountered in xDSL transmissions. Indeed, the upper frequency boundary of the present invention is defined in a range that is readily supported by, or compatible with, transmission systems (and attached POTS-type equipment) presently in place between a customer premises and a central office, without the need for extraneous devices such as POTS filters and POTS splitters.

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 (PSD) 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. **9**. When, however, there is a demand for voice transmissions, 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.

Illustrated in FIG **8** is the alternative embodiment of the present invention that achieves simultaneous multiple telephone type services on a single wire pair by utilizing the frequency division multiplexing method. Frequency division multiplexing assigns an available frequency range, within the band with the communication medium, for each device that is separately addressable. As shown in FIG. **8**, the POTS devices of the prior art telephone **44**, standard fax machine **42**, and the like, establish communications on a frequency range between 0 kHz and about 4 kHz as shown as item **111**. A second transmission frequency range defined at a higher frequency level **121** provides simultaneous multiple access for a service device. Each available frequency range within the bandwidth of the communication medium can be assigned to a particular service type. While FIG. **8** illustrates five frequency ranges **121** through **125**, the invention can utilize two or more frequency ranges between 20 kHz and 1 MHz.

Referring now to FIG. **10**, illustrated is the routine that initializes and processes the address determination logic within the DSP, CPU or ASIC **71** (FIG. **4**). Initialization of the address determination routine of the DSP, CPU or ASIC occurs at step **131**. This initialization step loads startup routines for the address determination logic. It is then determined if the address by the DSP, CPU or ASIC determination logic is performed by utilizing a multipoint protocol, which provides each device with a unique device ID that is uniquely and separately addressable, or if the address determination logic uses frequency division multiplexing, that is accomplished by assigning frequency ranges to each unique service device at step **132**. If it is determined at step **132** that a multipoint protocol with unique device addresses is being utilized, then step **134** sets the address determination logic to multipoint DSL and assigns the unique device IDs to the available service devices. If it is determined at step **132**, that frequency division multiplexing is to be utilized, then each service device is assigned a unique frequency range at step **133**. Step **135** starts processing communications for each of the assigned service devices. Processing continues until the service device is separated from the network and the address determination logic is exited at step **139**.

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 is claimed is:

1. A data communications apparatus for enabling a plurality of telephone-type services to be provided to a plurality of telephone-type devices connected to a single subscriber line, said apparatus comprising:

circuitry connectable to at least one of said plurality of telephone-type devices wherein said circuitry enables said device to operate using a frequency band higher than a POTS frequency band on the single subscriber line; and

interface circuitry attachable to said single subscriber line to enable said at least one of said plurality of telephone-type devices using said frequency band higher than said POTS frequency band to simultaneously share said single subscriber line with another one of said plurality of telephone-type devices that uses the POTS frequency band,

wherein each of said plurality of telephone-type services is identified by a unique telephone-type number.

2. The apparatus of claim **1**, wherein said plurality of services are concurrent.

3. The apparatus of claim **1**, wherein each of said plurality of telephone-type service is determinable by a telephone service provider.

4. The apparatus of claim **1**, wherein said data communications apparatus further provides digitized analog communications.

5. The apparatus of claim **1**, wherein said data communications apparatus further provides audio services utilizing audio compression.

6. The apparatus of claim **1**, wherein said data communications apparatus further provides DSL service.

7. The apparatus of claim **1**, wherein said data communications apparatus further provides Multimedia service.

8. The apparatus of claim **1**, wherein said data communications apparatus further provides video phone service.

9. The apparatus of claim **1**, wherein said data communications apparatus further provides digital phone service with a protocol required for communication over Internet.

10. The apparatus of claim **1**, wherein said data communications apparatus further provides digital facsimile service.

11. The apparatus of claim **1**, wherein said data communications apparatus further provides audio and visual data streaming service.

12. The apparatus of claim **1**, wherein said data communications apparatus further provides a home security service.

13. The apparatus of claim **1**, wherein said data communications apparatus further provides a meter reading system service.

14. A method for use in a data communications apparatus, said method comprising:

providing a plurality of telephone-type services to a plurality of telephone-type devices connected to a single subscriber line using a frequency band higher than a POTS frequency band; and

enabling said at least one of said plurality of telephone-type devices using said frequency band higher than said

POTS frequency band and attached to said data communications apparatus to simultaneously share said single subscriber line with another one of said plurality of telephone-type devices that uses the POTS frequency band,

wherein the step of providing a plurality of telephone-type services further comprises the step of identifying each of said plurality of telephone-type services by a unique telephone-type number.

15. The method of claim 14, wherein the step of providing a plurality of telephone-type services further comprises the step of:

providing said plurality of telephone-type services concurrently.

16. The method of claim 14, wherein the step of providing a plurality of telephone-type services further comprises the step of:

making each of said plurality of telephone-type services determinable by a telephone service provider.

17. The method of claim 14, further comprising the step of:

providing digitized analog communications.

18. The method of claim 14, further comprising the step of:

providing audio services utilizing audio compression.

19. The method of claim 14, further comprising the step of:

providing DSL service.

20. The method of claim 14, further comprising the step of:

providing digital phone service with a protocol required for communication over Internet.

21. The method of claim 14, further comprising the step of:

providing digital facsimile service.

22. The method of claim 14, further comprising the step of:

providing audio and visual data streaming service.

23. The method of claim 14, further comprising the step of:

providing a home security service.

24. The method of claim 14, further comprising the step of:

providing a meter reading system service.

25. The method of claim 14, further comprising the step of:

providing Multimedia service.

26. The method of claim 14, further comprising the step of:

providing video phone service.

27. A data communications system comprising:

a means for connecting to at least one of a plurality of telephone-type devices that uses a frequency band higher than a POTS frequency band; and

a means for providing a plurality of telephone-type services to said plurality of telephone-type devices connected to a single subscriber line, said providing means enabling said at least one of said plurality of telephone-type devices attached to said data communications system using a frequency band higher than a POTS frequency band to simultaneously share said single subscriber line with another one of said plurality of telephone-type devices that uses the POTS frequency band,

wherein said telephone-type services providing means further comprises a means for identifying each of said plurality of telephone-type services by a unique telephone-type number.

28. The system of claim 27, wherein said telephone-type services providing means further comprises:

a means for providing said plurality of telephone-type services concurrently.

29. The system of claim 27, wherein said telephone-type services providing means further comprises:

a means for making each of said plurality of telephone-type services determinable by a telephone service provider.

30. The system of claim 27, further comprising:

a means for providing digitized analog communications.

31. The system of claim 27, further comprising:

a means for providing audio services utilizing audio compression.

32. The system of claim 27, further comprising:

a means for providing DSL service.

33. The system of claim 27, further comprising:

a means for providing Multimedia service.

34. The system of claim 27, further comprising:

a means for providing video phone service.

35. The system of claim 30, further comprising:

a means for providing digital phone service, said service includes a protocol required for communication over Internet.

36. The system of claim 27, further comprising:

a means for providing digital facsimile service.

37. The system of claim 27, further comprising:

a means for providing audio and visual data streaming service.

38. The system of claim 27, further comprising:

a means for providing a home security service.

39. The system of claim 27, further comprising:

a means for providing a meter reading system service.

40. A method comprising the steps of:

providing a first telephone-type service over a subscriber line in a POTS frequency band,

providing a second telephone-type service over the subscriber line in a frequency band that is higher than said POTS frequency band; and

identifying each of the first and second telephone-type devices with a different telephone-type number.

41. The method of claim 40, wherein the providing steps are performed at a central office.

42. The method of claim 40, wherein the providing steps are performed at a customer premises.

43. The method of claim 40, further comprising the steps of:

communicating a data signal in the frequency band that is higher than the POTS frequency band; and

communicating a voice signal in the POTS frequency band.

44. The method of claim 40, further comprising the step of:

providing said first and second telephone-type services concurrently on the subscriber line.

45. The method of claim 40, further comprising the step of:

making each of the plurality of telephone-type services determinable by a telephone service provider.

46. The method of claim 40, further comprising the step of·

providing digitized analog communications in the frequency band that is higher than the POTS frequency band.

47. The method of claim 40, wherein the second telephone-type service is an audio service utilizing audio compression.

48. The method of claim 40, wherein the second telephone-type service is a DSL service.

49. The method of claim 40, wherein the second telephone-type service is a multimedia service.

50. The method of claim 40, wherein the second telephone-type service is video phone service.

51. The method of claim 40, wherein the second telephone-type service a digital phone service, the service including a protocol for communication over Internet.

52. The method of claim 40, wherein the second telephone-type service a digital facsimile service.

53. The method of claim 40, wherein the second telephone-type service an audio and visual data streaming service.

54. The method of claim 40, wherein the second telephone-type service a home security service.

55. The method of claim 40, wherein the second telephone-type service is a meter reading system service.

56. A central office wire center system, comprising:

means for providing a first telephone-type service over a subscriber line in a POTS frequency band;

means for providing a second telephone-type service over the subscriber line in a frequency band that is higher than the POTS frequency band; and

means for identifying each of the first and second telephone-type devices by a different telephone-type number.

57. The system of claim 56, wherein the first and second telephone-type services are provided concurrently on the subscriber line.

58. The system of claim 56, further comprising:

a means for making each of the first and second telephone-type services determinable by a telephone service provider.

59. The system of claim 56, wherein the second telephone-type service uses digitized analog communications.

60. The system of claim 56, wherein the second telephone-type service is an audio service utilizing audio compression.

61. The system of claim 56, wherein the second telephone-type service is a DSL service.

62. The system of claim 56, wherein the second telephone-type service is a multimedia service.

63. The system of claim 56 wherein the second telephone-type service is a video phone service.

64. The system of claim 56, wherein the second telephone-type service is a digital phone service, the service including a protocol required for communication over Internet.

65. The system of claim 56, wherein the second telephone-type service is a digital facsimile service.

66. The system of claim 56, wherein the second telephone-type service is an audio and visual data streaming service.

67. The system of claim 56, wherein the second telephone-type service is a home security service.

68. The system of claim 56, wherein the second telephone-type service is a meter reading system service.

69. A system, comprising:

a subscriber line;

one or more first telephone-type devices connected to the subscriber line and communicating in a POTS frequency band;

one or more second telephone-type devices connected to the subscriber line and communicating in a frequency band that is higher than the POTS frequency band; and

wherein the first and second telephone-type devices are identified by a respective telephone-type number.

70. The system of claim 69, wherein the existence of the first and second telephone-type services can be determined by a telephone service provider.

71. The system of claim 69, wherein at least one of the one or more second telephone-type devices is a digitized analog communications device.

72. The system of claim 69, wherein at least one of the one or more second telephone-type devices implements an audio service, utilizing audio compression.

73. The system of claim 69, wherein at least one of the one or more second telephone-type devices implements a DSL service.

74. The system of claim 69, wherein at least one of the one or more second telephone-type devices implements a multimedia service.

75. The system of claim 69, wherein at least one of the one or more second telephone-type devices implements a video phone service.

76. The system of claim 69, wherein at least one of the one or more second telephone-type devices implements a digital phone service, the service including a protocol for communication over the Internet.

77. The system of claim 69, wherein at least one of the one or more second telephone-type devices implements a digital facsimile service.

78. The system of claim 69, wherein at least one of the one or more second telephone-type devices implements an audio and visual data streaming service.

79. The system of claim 69, wherein at least one of the one or more second telephone-type devices implements a home security service.

80. The system of claim 69, wherein at least one of the one or more second telephone-type devices implements a meter reading system service.

81. The system of claim 69, wherein at least one of the second telephone-type devices has a transfer rate that is higher than that associated with at least one of the first telephone-type devices.

82. A system, comprising:

a subscriber line;

a first telephone-type device connected to the subscriber line;

a second telephone-type device designed to communicate in a frequency band above the POTS frequency band; and

interface circuitry that interfaces the second telephone type-device to the subscriber line and enables the second telephone-type device to communicate in a frequency band that is higher than the POTS frequency band,

wherein the first and second telephone-type devices are identified by a respective telephone-type number.

83. The system of claim 82, wherein at least one of the one or more second telephone-type devices is a digitized analog communications device.

84. The system of claim 82, wherein at least one of the one or more second telephone-type devices implements an audio service that utilizes audio compression.

85. The system of claim **82**, wherein at least one of the one or more second telephone-type devices implements a DSL service.

86. The system of claim **82**, wherein at least one of the one or more second telephone-type devices implements a multimedia service.

87. The system of claim **82**, wherein at least one of the one or more second telephone-type devices implements a video phone service.

88. The system of claim **82**, wherein at least one of the one or more second telephone-type devices implements a digital phone service, the service including a protocol for communication over the Internet.

89. The system of claim **82**, wherein at least one of the one or more second telephone-type devices implements a digital facsimile service.

90. The system of claim **82**, wherein at least one of the one or more second telephone-type devices implements an audio and visual data streaming service.

91. The system of claim **82**, wherein at least one of the one or more second telephone-type devices implements a home security service.

92. The system of claim **82**, wherein at least one of the one or more second telephone-type devices implements a meter reading system service.

93. The system of claim **82**, wherein at least one of the first telephone-type devices is designed to communicate a voice signal, and at least one of the second telephone-type devices is designed to communicate a data signal.

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PATENT NO. : 6,580,785 B2
DATED : June 17, 2003
INVENTOR(S) : Bremer et al.

Page 1 of 2

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Title page,
Item [*] Notice, delete the phrase "57 days" and substitute therefor -- 58 days --.
Item [56], **References Cited**, U.S. PATENT DOCUMENTS, add the patent
-- 5,970,473 10/1999 Gerszberg et al. 705/26 --.

Column 1,
Line 60, after the word "the", delete the word "instrumentality's" and substitute therefor
-- instrumentalities --.

Column 2,
Line 22, after the word "high-speed", delete the word "single" and substitute therefore
-- signal --.
Line 61, after the word "that", delete the word "provides".

Column 3,
Line 1, after the word "the", delete the word "Frequency" and substitute therefor
-- frequency --.
Lines 4-5, after the word "initializing", add the word -- of --.
Line 29, after the phrase "(PCs)", delete the numeral "43" and substitute therefor -- 46 --.
Line 35, after the word "office", add the phrase -- (CO) --.
Line 43, after the word "center", add the number -- 11 --.

Column 5,
Line 24, after the date "1998", add the phrase -- now issued as U.S. Patent No. 6,084,885, issue date of July 4, 2000, --.

Column 7,
Line 19, after the word "described", delete the phrase "in 3A".
Line 19, after the word "above", delete the phrase "as item 3X".
Line 57, after the word "communicated", delete the word "through".

UNITED STATES PATENT AND TRADEMARK OFFICE
CERTIFICATE OF CORRECTION

PATENT NO. : 6,580,785 B2
DATED : June 17, 2003
INVENTOR(S) : Bremer et al.

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 8,
Line 1, after the word "interfaces", delete the phrase "91I-91N" and substitute therefor -- 91A-91N --.
Line 2, after the word "lines", delete the phrase "27I-27N" and substitute therefor -- 27A-27N --.

Column 13,
Lines 15, 18, 20 and 23, after the first occurrence of the word "service", add the word -- is --.

Signed and Sealed this

Sixth Day of January, 2004

JAMES E. ROGAN
Director of the United States Patent and Trademark Office

EXHIBIT B
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US006546090B1

(12) **United States Patent**
Bremer et al.

(10) **Patent No.: US 6,546,090 B1**
(45) **Date of Patent: *Apr. 8, 2003**

(54) **APPARATUS AND METHOD FOR COMMUNICATING VOICE AND DATA BETWEEN A CUSTOMER PREMISES AND A CENTRAL OFFICE**

(75) Inventors: **Gordon Bremer**, Clearwater, FL (US); **Thomas Bingel**, Belleair Beach, FL (US)

(73) Assignee: **Paradyne Corporation**, Largo, FL (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

This patent is subject to a terminal disclaimer.

(21) Appl. No.: **09/374,774**

(22) Filed. **Aug. 16, 1999**

Related U.S. Application Data

(62) Division of application No. 08/962,796, filed on Nov. 3, 1997, now Pat. No. 6,061,392.

(60) Provisional application No. 60/033,660, filed on Dec. 17, 1996.

(51) **Int. Cl.^{7}** **H04M 11/00**

(52) **U.S. Cl.** **379/93.08**

(58) **Field of Search** 379/93.28, 93.31, 379/93.08, 93.09, 93.01; 375/222

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6,061,392 A * 5/2000 Bremer et al. 375/222

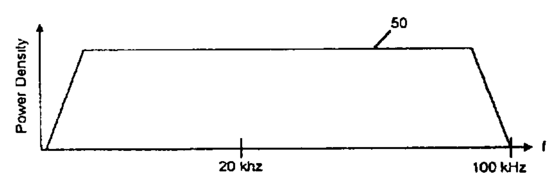
* cited by examiner

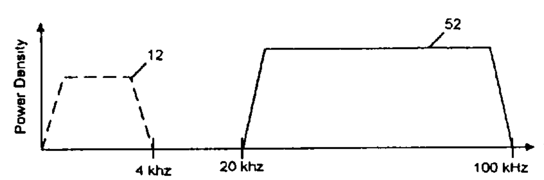
Primary Examiner—Stella Woo
(74) *Attorney, Agent, or Firm*—Thomas, Kayden, Horstemeyer & Risley, LLP

(57) **ABSTRACT**

A method and system are provided for communicating voice and data across a communication link, in a manner that senses and dynamically adapts to the simultaneous transmission of voice information across the local loop. In accordance with one aspect of the invention, a method is provided for dynamically communicating data over a local loop using a modem comprising the steps of transmitting data in a full-band transmission state, sensing a band-limiting condition, and adjusting the transmission of data from the full-band transmission state to a band-limited transmission state, in response to the sensing step. In accordance with the method, data may be transmitted by the modem across the local loop at the same time that voice information is communicated via telephone across the same local loop. A significant aspect of the present invention is the dynamic allocation of the data transmission bandwidth, whereby the invention senses a condition indicative of whether voice information is being communicated. If so, then the system shifts and/or narrows the data transmission bandwidth to allow for voice communications without interference from or with the data transmission. However, when no voice information is being communicated, the invention dynamically allocates the data transmission bandwidth to utilize at least a portion, if not all, of the frequency band otherwise used for communicating voice information.

83 Claims, 7 Drawing Sheets





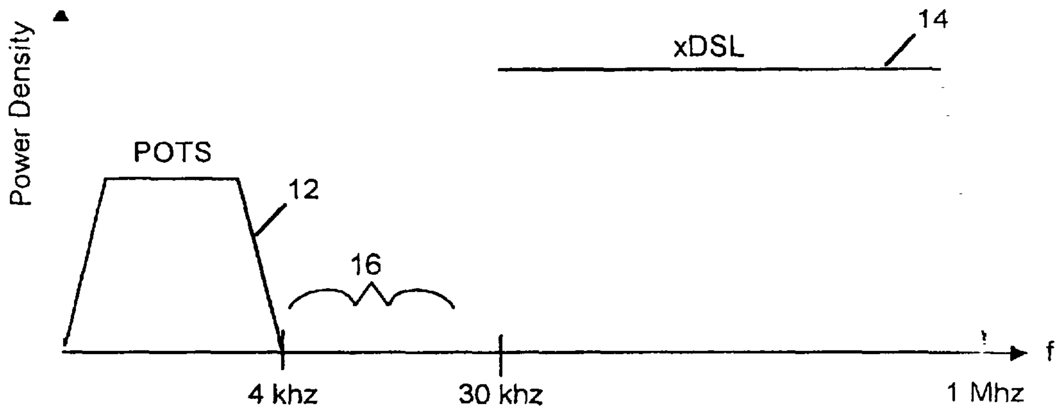


FIG. 1 (Prior Art)

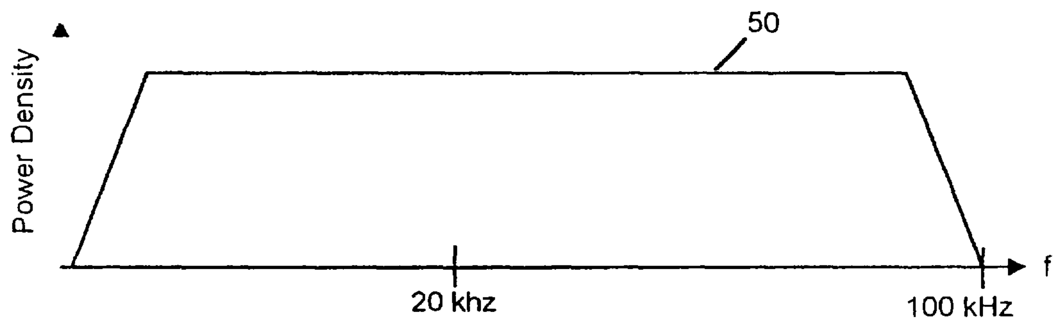


FIG. 3A

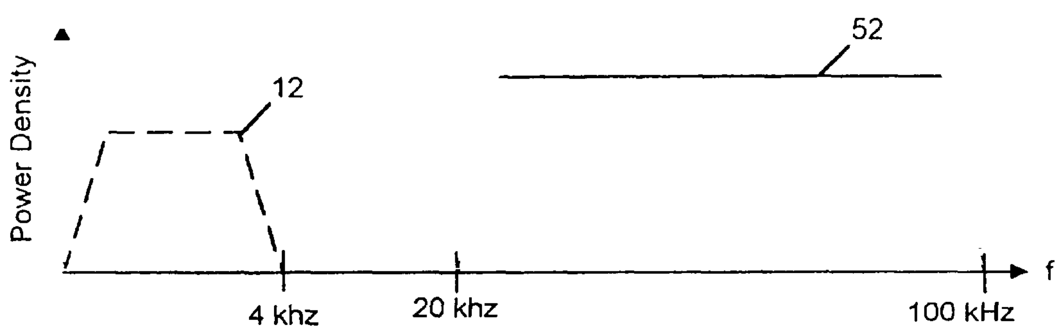


FIG. 3B

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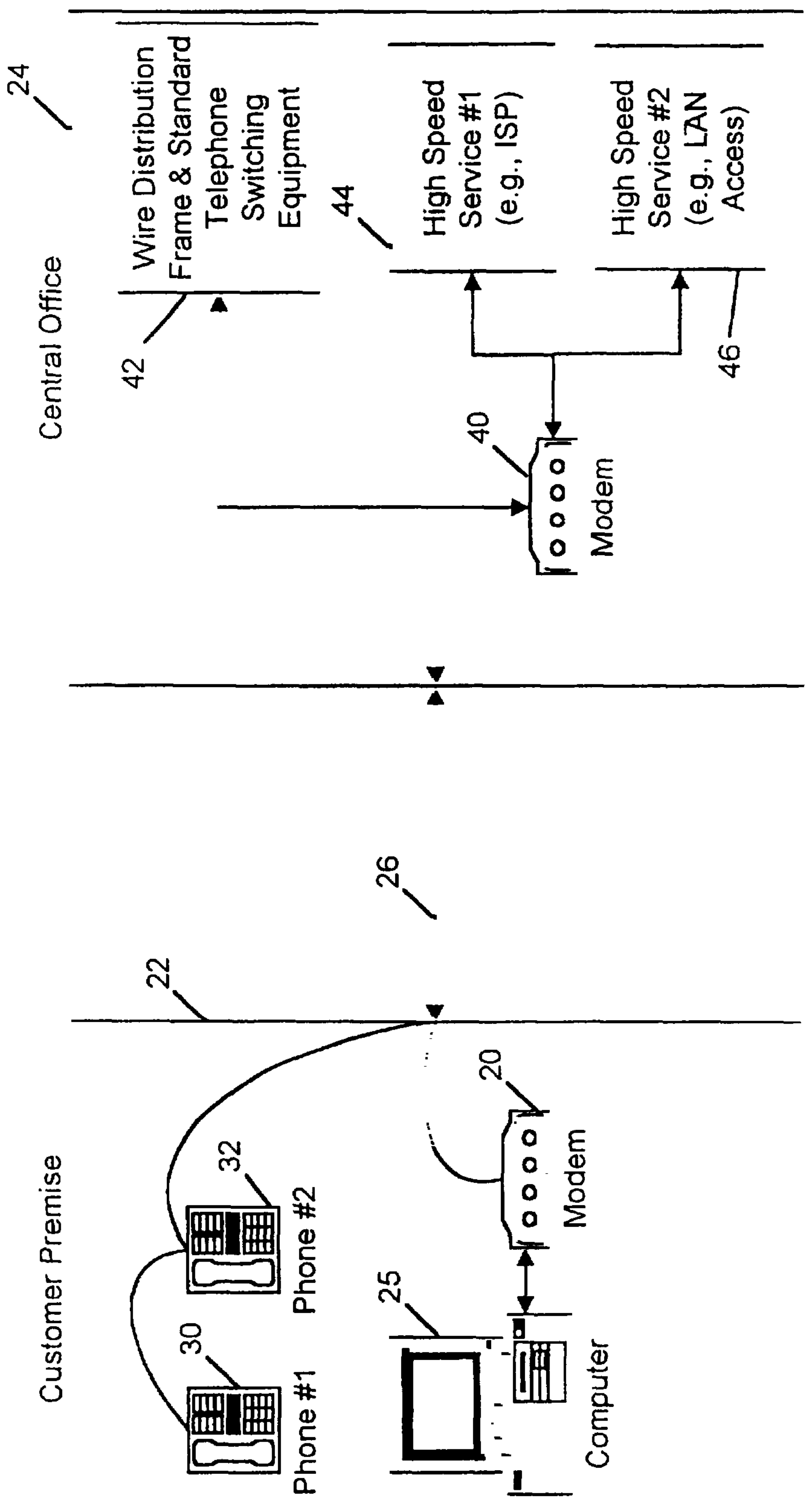
24
Central Office
Wire Distribution Frame & Standard Telephone Switching Equipment
42
44
High Speed Service #1 (e.g., ISP)
High Speed Service #2 (e.g., LAN Access)
46
40
Modem
26
22
Customer Premise
32
Phone #2
30
Phone #1
20
Modem
25
Computer

FIG. 2

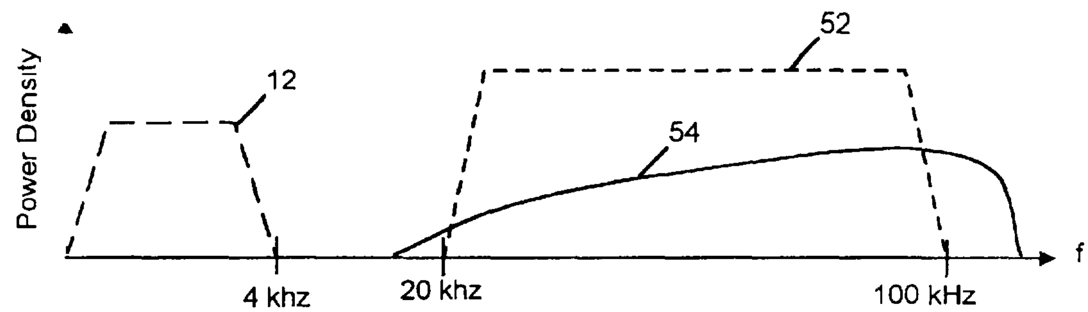

FIG. 3C

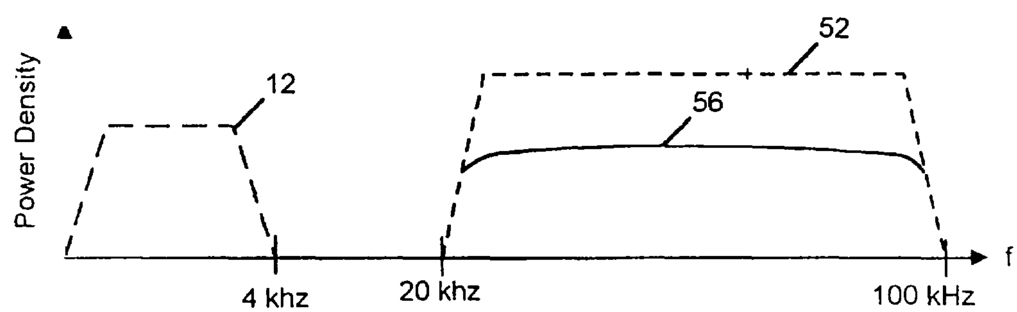

FIG. 3D

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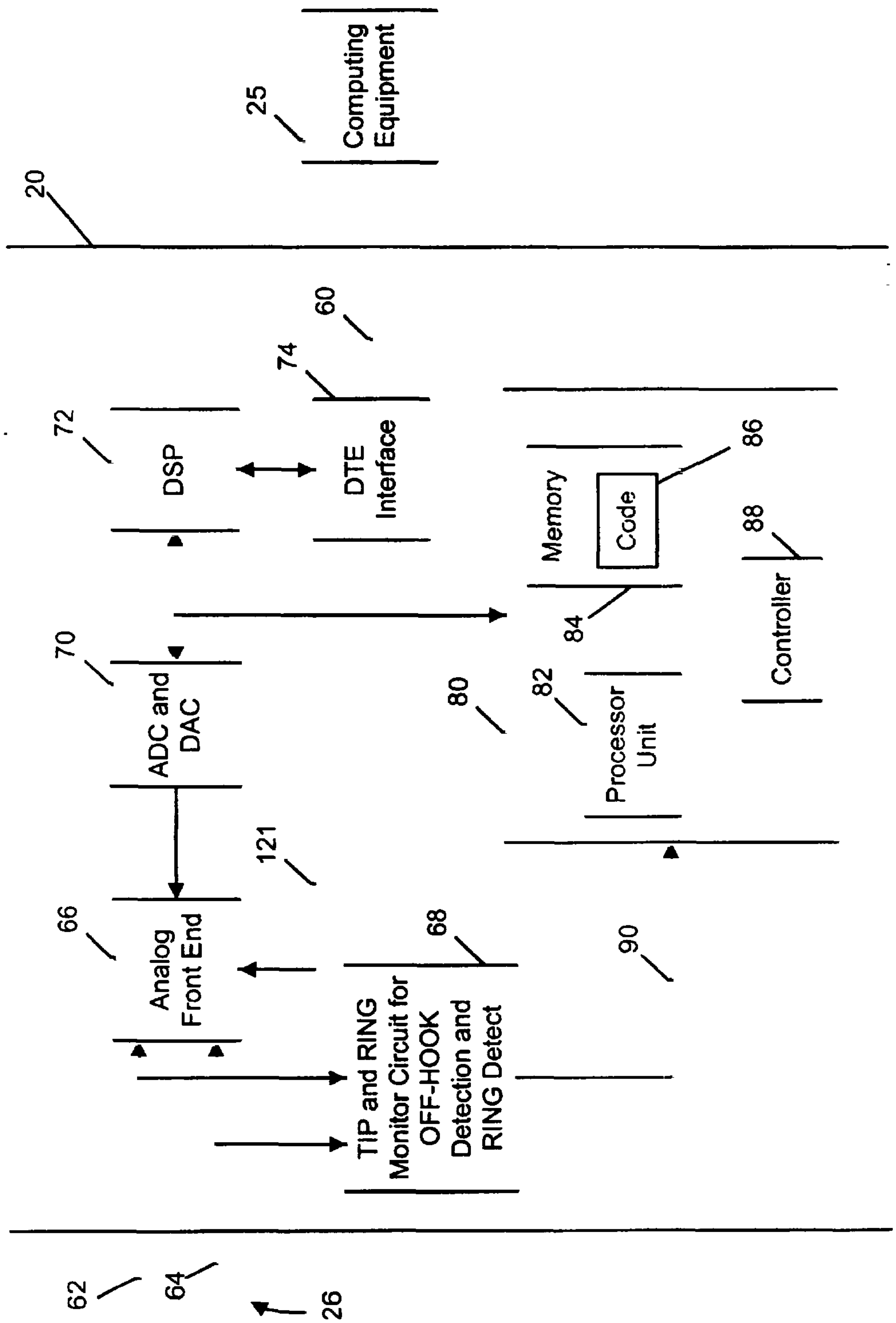
20
25
Computing Equipment
60
74
72
DSP
DTE Interface
Memory
Code
86
88
Controller
84
70
ADC and DAC
80
82
Processor Unit
121
66
Analog Front End
68
90
TIP and RING Monitor Circuit for OFF-HOOK Detection and RING Detect
62
64
26

FIG. 4

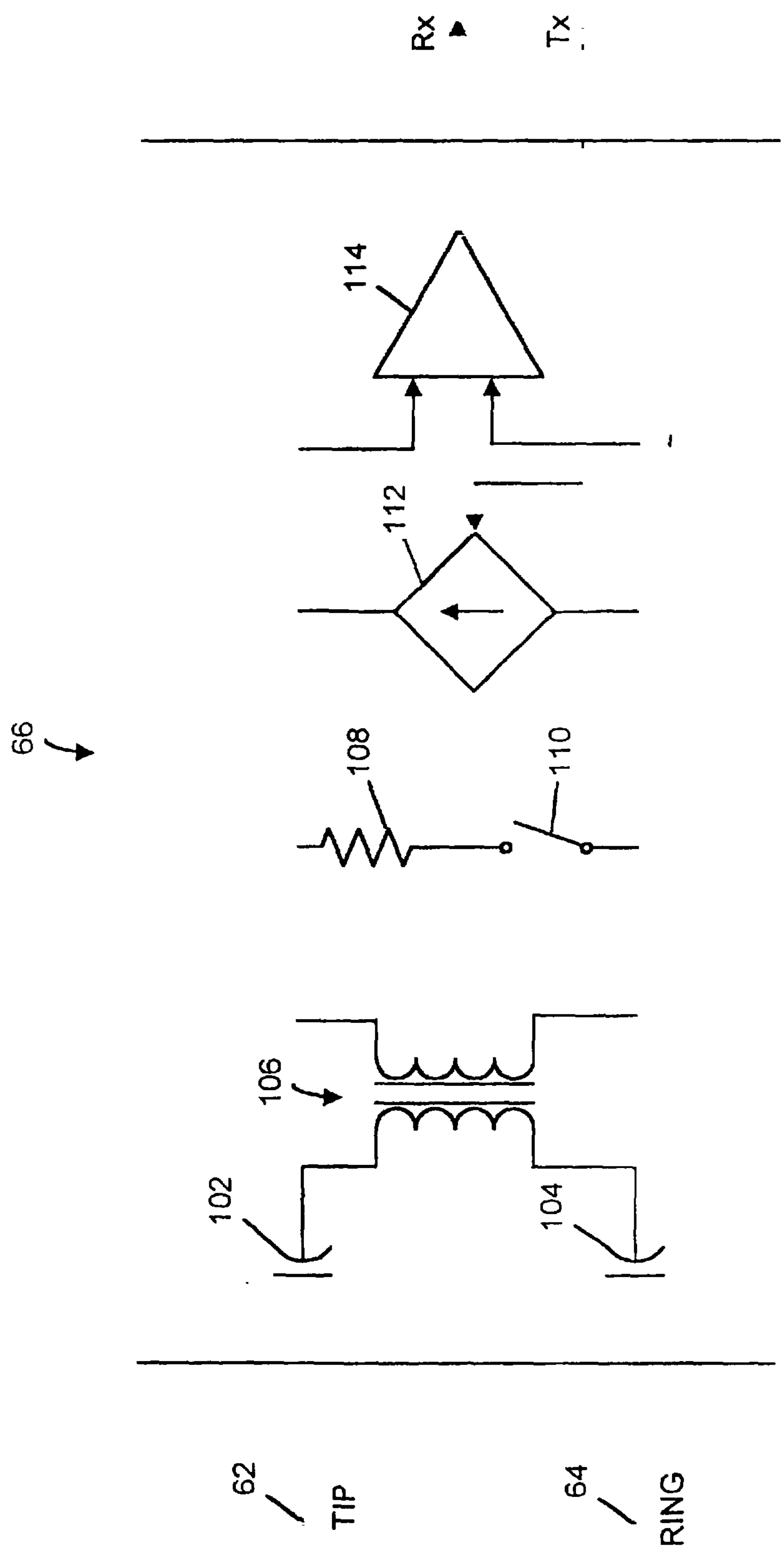

FIG. 5

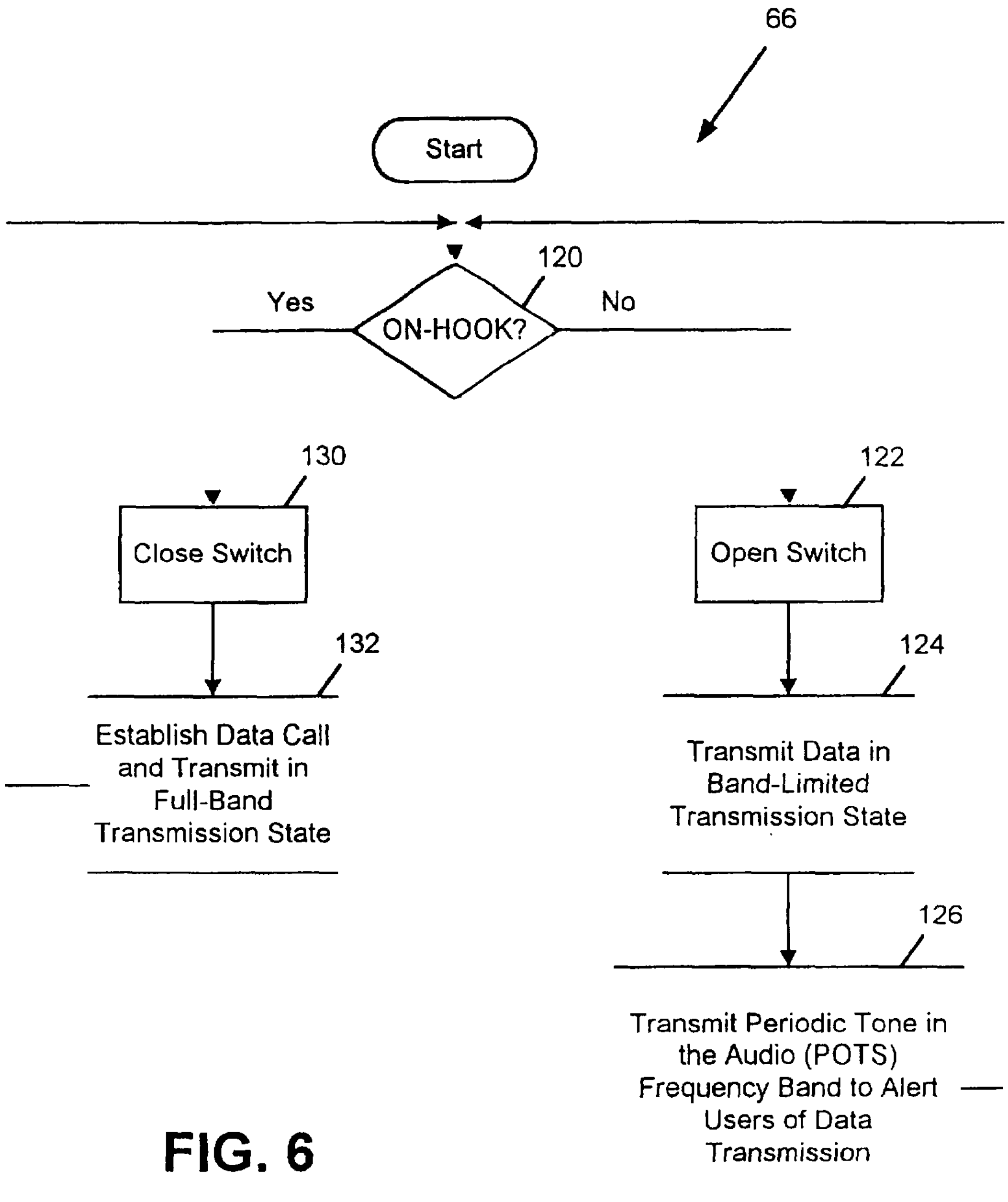
66
Start
120
Yes
ON-HOOK?
No
130
Close Switch
132
Establish Data Call and Transmit in Full-Band Transmission State
122
Open Switch
124
Transmit Data in Band-Limited Transmission State
126
Transmit Periodic Tone in the Audio (POTS) Frequency Band to Alert Users of Data Transmission

FIG. 6

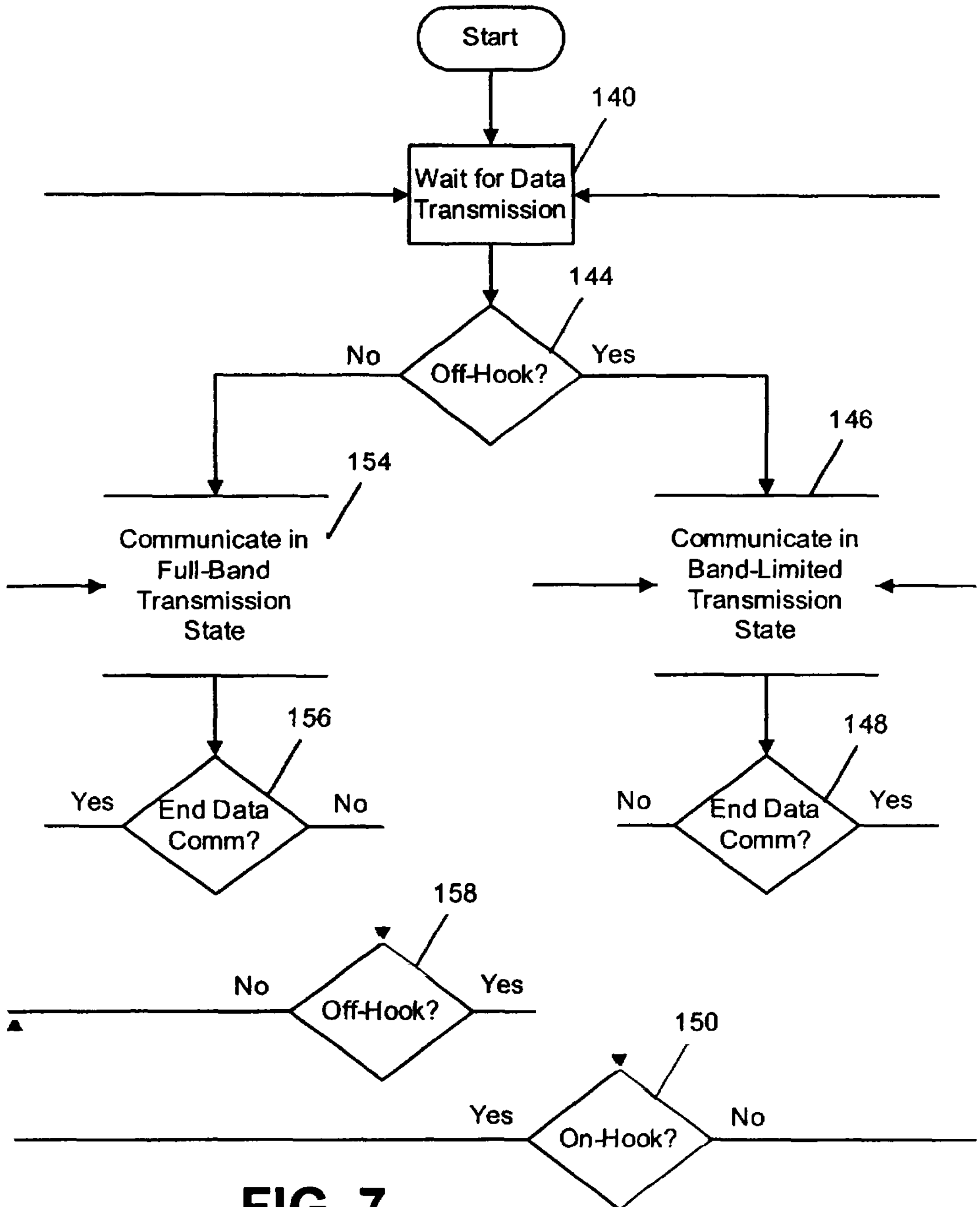

FIG. 7

EXHIBIT C
PAGE 8 OF 18

APPARATUS AND METHOD FOR COMMUNICATING VOICE AND DATA BETWEEN A CUSTOMER PREMISES AND A CENTRAL OFFICE

RELATED APPLICATION

This application claims the benefit of U.S. Provisional Patent Application Ser. No. 60/033,660, filed on Dec. 17, 1996, and entitled Digital Subscriber Loop Data Communications Method Enabling Simultaneous Data and POTS Without POTS Filters/Splitters or Special Premise Wiring. This application is a divisional of and claims the benefit of U.S. patent application Ser. No. 08/962,796, filed on Nov. 3, 1997, now issued as U.S. Pat. No. 6,061,392 and entitled APPARATUS AND METHOD FOR COMMUNICATING VOICE AND DATA BETWEEN A CUSTOMER PREMISES AND A CENTRAL OFFICE. The foregoing provisional application 60/033,660 and the foregoing issued U.S. Pat No. 6,061,392 are hereby incorporated herein by reference in their entirety.

FIELD OF THE INVENTION

The present invention generally relates to system for communicating both voice and data over modems, and more particularly to high speed modems offering robust communication between a central office and a customer premises.

BACKGROUND OF THE INVENTION

High speed digital modems, such as Rate Adaptive Digital Subscriber Loop ("RADSL") modems, are able to transfer data at high rates over the local loop, because they use frequencies which are significantly higher than the voice band frequencies used in Plain Old Telephone Service ("POTS"). By way of example, speech on a POTS system generally occurs in the frequency spectrum between about 0 Hz ("DC") and about 4 kHz, whereas RADSL modems use the frequency spectrum of between about 20 kHz to about 1 MHz. High speed digital modems generally include error detection circuitry which measures the errors which occur during communications. By making such measurements, they are then able to update their statistical knowledge of the wire pair which extends between the subscriber's location and the central office. Using that statistical knowledge, the modems can select optimal operating speeds. These modems were originally proposed when it was thought that services, such as video-on-demand, would be desirable.

As modem technology has developed, another need has arisen, in that the Internet has become a popular medium for both personal and work related use.

While the high speeds of RADSL modems seem to be quite desirable, their use of high frequencies mean that they also need to be protected from high frequency noise, such as cross-talk from adjacent channels or adjacent loops in the loop cable binder, as such noise causes them to downwardly adjust their operating speeds. In order to avoid certain types of noise, RADSL modems typically require the use of filters, called POTS filters, together with splitters for isolating Public Switched Telephone Network ("PSTN") equipment from the RADSL modems. Indeed, without POTS filters and POTS splitters, POTS signals directly interfere with the RADSL spectrum below about 20 kilohertz and the RADSL spectrum directly interferes with the POTS. POTS filters and POTS splitters reduce POTS signaling transients from interfering with RADSL data transmission. In addition, the use of the high RADSL bandwidth demands relatively high transmit power, which can cause distortions and dynamic range overload to POTS equipment.

Unfortunately, the manufacture and installation of POTS filters and splitters are expensive, and their use sometimes requires rewiring of the customer premises to ensure that all PSTN equipment is properly isolated from the RADSL modems and computing equipment. Consequently, it would be desirable to avoid the use of POTS splitters and filters, in order to avoid the expense they impose (e.g., purchase cost and possible rewiring of customer premises).

Accordingly, there appears to be a need for a mass market modem which has data transfer rates greater than the 33.6 Kbps attainable by PSTN modems, yet under the rate that requires the addition of POTS filters, splitters, etc. to address noise and deleterious transmission line effects often encountered in high speed DSL modems.

Yet another problem which is manifest in increased Internet access and data communications is the increasingly limited availability to the customer phone line or local loop for its original purpose, i.e., voice communications. Of course, one solution is for a customer to purchase an additional phone line. This, however, imposes an additional cost on the customer. Moreover, unless the line is dedicated by the customer for a specific purpose (which is poor utilization), the second line may not always be available when needed.

Accordingly, there is a need to provide an improved modem that accommodates data transmissions, while simultaneously allowing traditional voice operation of a telephone attached to the same line at the customer premise. It is particularly desirable to have such a modem that does not require the use of costly POTS filters and splitters.

SUMMARY OF THE INVENTION

Certain objects, advantages and novel features of the invention will be set forth in part in the description that follows and in part will become apparent to those skilled in the art upon examination of the following or may be learned with the practice of the invention. The objects and advantages of the invention may be realized and obtained by means of the instrumentalities and combinations particularly pointed out in the appended claims.

To achieve the advantages and novel features, the present invention is generally directed to a method and apparatus for communicating data across a local loop, in a manner that senses and dynamically adapts to the simultaneous transmission of POTS (e.g., voice or PSTN modem) information across the local loop. In accordance with one aspect of the invention, a method is provided for dynamically communicating data over a local loop using a modem comprising the steps of transmitting data in a full-band transmission state, sensing a band-limiting condition, and adjusting the transmission of data from the full-band transmission state to a band-limited transmission state, in response to the sensing step. The step of sensing a band-limiting condition includes both the detection of the onset of a condition indicating that the method should enter the band-limited transmission state, as well as the detection of the cessation of that condition, indicating that the method should enter the full-band transmission state from the band-limited transmission state.

In accordance with the method of the present invention, data may be transmitted by the modem across the local loop at the same time that POTS (e.g., voice or PSTN modem data) information is communicated across the same local loop. A significant aspect of the present invention is the dynamic allocation of the data transmission bandwidth,

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whereby the invention senses a condition indicative of whether POTS information is being communicated. If so, then the system shifts and/or narrows the data transmission bandwidth to allow for voice communications without interference from or with the data transmission. However, when no POTS information is being communicated, the invention dynamically allocates the data transmission bandwidth to utilize at least a portion, if not all, of the frequency band otherwise used for communicating voice information.

In accordance with the preferred embodiment, the method senses an off-hook condition of a telephone handset of a telephone electrically connected to the local loop. In use, a local loop extending between a customer premises and a central office branches, at the customer premise, to support multiple connections to the local loop. In this regard, the various branches or connections are typically routed throughout a customer premises to phone jacks, such as RJ-11 jacks. Multiple telephones may be plugged directly into these jacks for voice communication across the local loop. Similarly, a modem constructed in accordance with the present invention may be plugged directly into one of these jacks. The off-hook condition is preferably sensed by detecting either a change in impedance in the telephone line, or alternatively, a drop in line voltage across the telephone line.

In accordance with one embodiment of the invention, the full-band transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of less than about 15–20 kilohertz (and preferably less than 4 kilohertz). In the band-limited transmission state, the transmission frequency bandwidth has a lower frequency boundary of greater than 4 kilohertz. The significance of these values, for purposes of the invention, is that when no voice information is being communicated across the local loop, the transmission frequency bandwidth invades that frequency band generally dedicated to the transmission of voice information (i.e., the 0–4 kilohertz POTS frequency band). When, however, the invention senses that POTS information is being communicated across the local loop, or that there is a demand for the POTS band (e.g., telephone off-hook, ring, etc.), then the embodiment shifts the lower boundary of the transmission frequency bandwidth above the generally 4 kilohertz upper limit of the voice band. Preferably, the lower boundary will be shifted upwardly to approximately 20 kilohertz, to allow sufficient separation between the voice and data transmission frequency bands so that no interference between the two is realized, either by voice information corrupting data, or data transmission being heard in the voice band as noise.

For purposes of the preferred embodiment of the present invention, the precise value of the upper boundary of the transmission frequency bandwidth is not so significant, as it is the dynamic adjustment of the lower boundary and/or the reduced power in POTS mode, that realizes the inventive step. However, it will be appreciated that the upper boundary will generally be greater than 40 kilohertz in order to define a meaningful transmission frequency bandwidth for data transmission. Indeed, in the preferred embodiment, the upper frequency boundary is approximately 80 kilohertz. It is believed that this frequency is low enough that transmissions may be effectively implemented without the need for POTS filters or POTS splitters, and therefore significantly reducing the cost of implementing the inventive system. Signal-to-noise ratio is high to permit reasonable data throughput without excessive power incident on attached POTS devices. Also, premises wiring and subscriber loop stubs do not cause substantive nulls in the frequency response. It will be further appreciated that shifting of the

upper frequency boundary is not relevant to the present invention. That is, the upper boundary may be shifted in conjunction with the shifting of the lower frequency boundary, or alternatively, the upper frequency boundary may remain substantially fixed.

It will be further appreciated that depending upon loading, line conditions, and other factors the spectral shape of the band-limited xDSL transmission may be varied to minimize noise, intermodulation products, or other interference within the POTS frequency band. More particularly, it is generally understood that the power density of xDSL transmissions is generally greater than that of POTS transmissions. Merely shifting the xDSL transmission into the band-limited transmission state with a lower cutoff frequency of approximately 20 kHz may not always provide a wide enough guard band to prevent interference with the POTS band. Line loading, line conditions, and other factors (which differ among local loops) factor into this determination. Intermodulation products are another source of noise that often is present within the POTS band. When such noise is present within the POTS band, the band-limited transmission state may be further configured by reducing the power-density of the xDSL transmission. Another, related solution may be to uniquely shape the spectral curve for xDSL transmissions. This, for example, may be done by tapering the lower frequency portion of the curve (i.e., that portion near the approximately 15–20 kHz frequency).

In accordance with another aspect of the preferred embodiment, a modem is provided for communicating data across a local loop. The modem includes an input/output signal line that is electrically connected with the local loop (e.g., plugged into an RJ-11 phone jack). The modem also includes a processor unit that is adapted for operation in one of two states: a full-band transmission state and a band-limited transmission state. The full-band transmission state is defined by a lower frequency boundary at a value below approximately 15–20 kilohertz and an upper frequency boundary generally greater than 40 kilohertz (as discussed above). The band-limited state is defined by a lower frequency boundary greater than 4 kilohertz and an upper frequency boundary greater than 40 kilohertz (which may or may not be the same as the upper frequency boundary for the full-band transmission state). The modem further includes a sensor or other sensing means for sensing that the local loop is in POTS mode (e.g. transmitting POTS information, or preparing to transmit POTS information), and the data signal power and bandwidth are adoptively altered to provide data without out inteferring with the POTS transmission. Upon sensing the band-limiting condition, such as an off-hook condition, the controller causes the processor unit to upwardly shift the lower frequency boundary of the transmission frequency band and operate in the band-limited, or reduced-power, state. Likewise, upon sensing no band-limiting condition (or a cessation in the band-limiting condition), the controller causes the processor unit to downwardly shift the lower frequency boundary of the transmission frequency band, and operate in the full-band transmission state, to maximize data throughput.

In accordance with yet a further aspect of the present invention, a method is provided for simultaneously communicating both voice and data between a customer premises and a central office across a local loop. In accordance with this aspect of the invention, the method comprises the steps of (1) transmitting data between the customer premises and the central office in a first frequency band, wherein the first frequency band is defined by an upper frequency boundary and a lower frequency boundary; (2) allocating a second

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frequency band for transmitting voice information between the customer premises and the central office; (3) sensing a band-limiting condition; and (4) dynamically shifting the lower frequency boundary of the first frequency band in response to the sensed band-limiting condition. In accordance with the invention, the lower frequency boundary of the first frequency band shifted to at least partially overlap the second frequency band when no band-limiting condition exists. The lower frequency boundary of the first frequency band is further shifted to avoid overlapping with any portion of the second frequency band when the band-limiting condition exists.

In accordance with yet a further aspect of the invention, a modem is provided for communicating across a communication link capable of single-use transmissions and multiple-use transmissions. The term single-use transmissions is used to generally connote that a single transmission or communication is occurring across the link. For example, a single PSTN voice call, or a single data communication transmission. The term multiple-use transmissions is used to generally imply that multiple transmissions or communications are occurring simultaneously. For example, the simultaneous transmission of a data communication and a PSTN voice call. The modem constructed in accordance with this aspect of the invention includes an input/output signal line in communication with the communication link. It further includes a processor unit adapted for operation in one of at least two states, a full-band transmission state and a band-limited state, wherein the full-band transmission state occurs when single-use transmissions are occurring across the transmission link, and the band-limited transmission state occurs when multiple-use transmissions are occurring across the communication link.

It will be appreciated that, in accordance with a broad inventive aspect, the present invention operates by adjusting transmit power between a band-limited transmission state and a full-band transmission state. Generally (but not necessarily always), the full-band transmission state occurs when the communication link is operating in a single-use transmission mode, while the band-limited transmission state generally occurs when the communication link is operating in a multiple-use transmission mode. In accordance with this broad concept of the invention, substantial transmission energy is transmitted by the modem in or near the POTS frequency band, when the modem is transmitting in the full-band state. Conversely, very little (ideally zero) energy is transmitted by the modem in or near the POTS frequency band, when the modem is transmitting in the band-limited state. This allows for simultaneous POTS transmissions (e.g., voice, PSTN modem, etc) in the POTS frequency band, and band-limited modem transmissions.

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 an illustration of the frequency spectrum of a dual frequency band communications system of the prior art, depicting the POTS transmission frequency band and the xDSL transmission frequency band;

FIG. 2 is a block diagram illustrating the primary components in a system utilizing the present invention;

FIG. 3A is a frequency spectrum illustrating the full-band transmission frequency band of the present invention;

FIG. 3B is a frequency spectrum illustrating the band-limited transmission frequency band of the present invention;

FIG. 3C is a frequency spectrum illustrating a band-limited transmission frequency band of an alternative embodiment of the present invention, having a uniquely shaped xDSL transmission band;

FIG. 3D is a frequency spectrum illustrating a band-limited transmission frequency band of an alternative embodiment of the present invention, having a reduced power xDSL transmission band;

FIG. 4 is a block diagram illustrating the primary components of a modem constructed in accordance with the present invention;

FIG. 5 is a circuit diagram illustrating the analog front end component of the modem block diagram of FIG. 4;

FIG. 6 is a software flowchart depicting the functional operation of the analog front end element, illustrated in FIG. 5; and

FIG. 7 is a software flowchart illustrating the top-level operation of a system constructed in accordance with the present invention.

DETAILED DESCRIPTION

Having summarized the invention, 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.

Turning now to the drawings, FIG. 1 is a diagram illustrating frequency band communications, as is known in the prior art. The term frequency band communications is used to indicate communication of information within a certain defined frequency band. As is known in the prior art, plain old telephone system (POTS) communications are transmitted in the frequency band 12 defined between about 0 (DC) and about 4 kHz. A second transmission frequency band 14 is defined at a higher frequency level than the POTS frequency band 12, and is used in the transmission of digital subscriber line (DSL) communications. A guard dead band 16 is typically provided to separate the two transmission frequency bands 12 and 14. The DSL transmission frequency band 14 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 14 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.

As will be discussed in more detail below, the present invention provides an upper transmission band having an upper frequency boundary that is much lower than the 1 MHz frequency boundary often encountered in xDSL transmissions. Indeed, the upper frequency boundary of the present invention is defined in a range that is readily supported by, or compatible with, transmission systems (and attached POTS-type equipment) presently in place between a customer premises and a central office, without the need for extraneous devices such as POTS filters and POTS splitters. In this regard, reference is made to FIG. 2, which is a top level diagram illustrating the principal hardware

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components of a system utilizing the present invention. In accordance with one aspect of the invention, a modem **20** is provided for achieving efficient data communications between a customer premises **22** and a central office **24** across a local loop **26**, by dynamically allocating a transmission frequency bandwidth and/or power 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 and/or power spectral density (PSD) 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. When, however, there is a demand for voice transmissions, then the present invention reallocates the transmission frequency band and PSD for the data communications so that there is no overlap or interference with the POTS transmission frequency band **12**, and so that there is not significant interference to POTS-type attached equipment.

In keeping within the description of FIG. **2**, the customer premises **22** may be a single-family household having a single phone line **26** for communicating between the customer premises **22** and a central office **24**. Within the house or customer premises **22**, multiple connections branch off of the local loop **26** and are terminated at phone jacks (such as RJ-11) located in various rooms of the household. In this way, multiple telephones **30** and **32** may be plugged in and supported from the same phone line **26**. In the same way, a personal computer **25** may be disposed in communication with the local loop **26** by way of a modem **20**.

Presently, unless a user purchases an additional phone line, or a more costly communication service, such as xDSL, simultaneous transmissions of voice and data to different locations are not possible. As a result, one person in a household may have the local loop **26** tied up with data communications (such as Internet communications), while another person at the same household is awaiting the use of the local loop **26** for voice communication. In accordance with the present invention, and as will be discussed in more detail below, this shortcoming is overcome.

In keeping with the description of FIG. **2**, a companion modem **40**, that is compatible with the modem **20**, is provided at the central office **24**. As is known, other equipment, such as wire distribution frame and standard telephone switching equipment **42** may also be in communication with the local loop **26**. Since the configuration and operation of such equipment is known in the prior art and does not effect or impact the present invention, it will not be discussed herein. FIG. 2 also illustrates a variety of services that may be connected at the central office **24** to the modem **40**, constructed in accordance with the present invention. These services may include a high speed ISP service **44**, a high speed LAN access service **46**, etc. Again, since the provision and operation of such services are generally understood and are further not necessary in order to describe the operation of the present invention, they will not be described herein.

Turning now to FIGS. **3A** and **3B**, the dynamic allocation and deallocation of the data transmission frequency band is illustrated. Specifically, FIG. **3A** illustrates the data transmission frequency band **50** in a full band transmission frequency state, while FIG. **3B** illustrates a data transmission frequency band **52** in a band-limited (POTS compatible) transmission frequency state. As illustrated in FIG. 3A, the full-band transmission frequency band **50** extends from approximately 0 Hz (DC) to approximately 100 kHz. In contrast, in FIG. **3B** the data transmission frequency band **52** extends from approximately 20 kHz to approximately 100 kHz. In accordance with an important aspect of the preferred embodiment, a modem **20** constructed in accordance with the invention senses the need to dynamically allocate or deallocate a portion of the transmission frequency band in order to accommodate voice communications within the 0 to 4 kHz POTS frequency band **12**. As will be described further herein, the present invention may sense this demand for voice transmissions (or band-limiting condition) by sensing an OFF-HOOK condition of a telephone **30, 32**, (see FIG. **2**) connected to the local loop **26**. Alternatively, this band-limiting condition may be detected by an impedance change on the local loop **26**.

For phone compatibility, in addition to detecting RING and OFF-HOOK conditions, the system may also be configured to detect voice conversation. Upon voice detection, the system may increase transmit power as it shifts into the band-limited transmission state, to increase data rate dynamically, so long as the voice band SNR is about 30 to 40 dB. When silence is once again detected (for a predetermined amount of time), the system will again reduce the transmit power for good idle channel perception.

Unlike typical xDSL communications, where the data transmission frequency band is often 1 MHz in width, the data transmission frequency band of the present invention is much less than that. This permits relatively high-speed data communication without the addition of expensive equipment, such as POTS splitters and POTS filters. Importantly, this addresses a market need from consumers that do not wish to incur, or cannot afford, the additional expenses normally incurred with purchasing an xDSL communication service. An important aspect of the present invention is its ability to sense when voiceband communications are not occurring, or otherwise when a band-limiting condition is not present, and expand the transmission frequency band into the frequency band otherwise reserved for POTS transmissions, and/or increase transmit power to increase the data rate. As can be seen from the illustrations in FIGS. 3A and 3B, expanding the transmission frequency band from a 20 kHz cutoff (FIG. 3B) to approximately DC (FIG. 3A) realizes a 25 percent increase in bandwidth (i.e., from 80 kHz to 100 kHz), and thus, realizes a significant improvement in performance.

FIGS. 3C and 3D illustrate alternative embodiments of the present invention. In short, FIGS. 3C and 3D illustrate a spectrally-shaped transmission curve and an adaptive power transmission curve, respectively. As illustrated in FIG. 3B, under normal operating condition, the power density of the xDSL transmission band is greater than that of the POTS transmission band. However, there may be instances when the guard band **16** is not large enough to sufficiently separate the xDSL transmission band **52** from the POTS frequency band **12**. As a result, xDSL transmissions may be evident in the POTS frequency band **12** as noise (audible static). The reasons this may occur are varied, and include factors such as telephone set sensitivity and non-linearities. Intermodulation products may also be manifest within the POTS transmission band **12** as noise.

It will be appreciated that, consistent with the concepts and teachings of the present invention, various adaptations of the band-limited transmission state may be implemented to minimize or eliminate noise in the POTS transmission band 12. One solution is to further increase the size of the guard band 16, thereby increasing the frequency separation between the POTS transmission band 12 and the xDSL transmission band 52. Another solution is to adaptively reduce the transmit power of the xDSL transmission band. This solution is illustrated in FIG. 3D, wherein the normal power spectrum 52 is illustrated in dashed line and the reduced power spectrum 56 is superimposed in solid line. Reducing the transmit power in this way reduces the amount of noise that is manifest within the POTS frequency band. The specific amount of power reduction may vary among customer premises, based upon the attached equipment.

Yet another solution is to more particularly define the spectral shape of the transmission band. This solution is illustrated in FIG. 3C. As shown, the power spectrum of the xDSL transmission band 54 may be asymmetrically shaped to provide a greater taper on the lower frequency end of the curve. This taper ensures sufficient attenuation of the xDSL transmission signal above the POTS frequency band 12, and therefore minimizes intermodulation products and noise (resulting from the xDSL transmission) within the POTS band 12. Although only one such shaped signal band 56 is illustrated in FIG. 3D, it will be appreciated that this aspect of the invention is not so limited. Instead, other shapes may be deemed desirable, depending upon the specific environment and line conditions.

Reference is now made to FIG. 4, which shows a block diagram of a modem 20 constructed in accordance with the present invention. As is common among modems, the modem 20 is in communication with both a local loop 26 and computing equipment 25, such as a personal computer. More specifically, the modem 20 communicates with the computing equipment 25 across line 60. The telephone line 26 is typically comprised of a two wire service, which wires are often denoted as TIP 62 and RING 64. The TIP 62 and RING 64 lines are input to an analog front-end circuit 66 (see FIG. 5) as well as a monitor circuit 68, which is configured to detect an OFF-HOOK condition of the local loop 26.

Analog-to-digital and digital-to-analog converter (ADC and DAC, respectively) circuitry 70 is in communication with the analog front end circuitry 66, and is in further communication with digital signal processor 72. Data received from the local loop 26 passes through the analog front-end 66 and is converted from analog-to-digital form by the analog to digital converter of block 70, before being passed to the digital signal processor 72. Conversely, outgoing data output from the digital signal processor 72 is converted by the digital to analog converter of block 70, before being communicated to the local loop 26, by way of the analog front-end 66. Finally, Data Terminal Equipment (DTE) interface 74 is in communication with the digital signal processor 72 and in further communication across line 60, with the data terminal equipment, such as a computer 25. The analog-to-digital and digital-to-analog converter circuitry 70, the digital signal processing 72, and the DTE interface 74 are all well known and generally operate in accordance with the prior art. Therefore, their individual structure and operation need not be described herein.

Indeed, a significant component of the modem 20, constructed in accordance with the present invention, is a controller 80 that is in communication with the various other components of the modem 20. While there are various ways to implement the controller 80, one way, as illustrated, is to further partition the controller 80 into functional units denoted as a processing unit 82, a memory 84 (which may further include an executable code segment 86), and a controller 88.

For purposes of the broad concepts of the present invention, the controller 80 receives a signal from the monitor circuit 68 on line 90, which signal indicates whether the invention should transmit data in a band-limited transmission state or a full-band transmission state. In this regard, the monitor circuitry 68 may be configured to detect an OFF-HOOK condition, or alternatively, a RING condition on local loop 26. As is known in the art, the OFF-HOOK condition may be detected by a drop in voltage across the local loop 26, or alternatively, a sudden change in impedance on the local loop 26. On the other hand, a RING detect condition is identified by a low frequency oscillatory voltage on local loop 26. For example, the voltage drops from about 48 volts (on hook) to approximately 10 volts or less (off hook), at the customer premises end of the local loop. In short, the controller 80 evaluates the signal received on line 90 to determine whether data should be transmitted in the full-band transmission state or the band-limited transmission state. Appropriate signals may, accordingly, be transmitted to the digital signal processor 72 for formulating data transmissions (or interpreting received data transmissions).

In accordance with an alternative embodiment of the invention, it will be appreciated that the monitor circuitry 68 may be incorporated within the controller 80, whereby certain signal conditions may be evaluated to detect the band-limiting condition. In this regard, an analog-to-digital converter 70 would also be implemented as part of the controller 80, to generate a signal in digital format which may be more readily evaluated and processed by the processing unit 82. In this regard the processing unit 82 may be a microprocessor, a microcontroller, an application specific integrated circuit (ASIC), or other digital circuitry configured to specifically process information. In the illustrated embodiment, the controller 80 includes fundamental components (processor unit 82, controller 88, memory 84) that together operate to perform distinct computing operations. Such operations may be controlled, for example, by executable code 86 contained within the memory 84.

Reference is now made to FIG. 5, which shows a more detailed diagram of the circuitry comprising the analog front end element 66. The preferred embodiment includes blocking capacitors 102 and 104, which are series connected with the TIP 62 and RING 64 signal lines, and serve to block any DC voltage otherwise carried on the TIP 62 and RING 64 lines. A transformer 106 couples alternating current to the remainder of the circuitry, as well as provides safety and signal isolation for the remaining circuitry in the modem. A termination resistor 108 and switch 110 are disposed for series connection with each other (depending upon whether the switch 110 is opened or closed), and together are connected in parallel across the secondary winding of the transformer 106. The switch 110 is controlled by controller 80 (FIG. 4) to close and therefore switch in the terminating resistor 108 when the telephones 30 and 32 (see FIG. 2) are all ON-HOOK (as observed by the monitor circuit 68). The switch 110 may be open to switch out the terminating resistor 108, upon detection of an incoming RING signal or OFF-HOOK on the local loop 26. Capacitors 102 and 104 are chosen to pass data, block DC, and yield acceptable Ringer Equivalence Number per FCC part 68. The switch 110 is generally opened to switch out the terminating resistor when the monitor circuit 68 determines that the local loop 26 is in the OFF-HOOK state. The reason for this is that, when one or more telephones are taken OFF-HOOK, then the OFF-HOOK telephone will terminate the line, and the terminating resistor 108 is not needed. Optionally, the switch 110 can be closed in the OFF-HOOK state to improve line termination provided by the OFF-HOOK telephone.

The item represented by reference numeral **112** denotes circuitry that is configured in a form of a dependent current source. The current source is prompted by the transmit signal Tx to create an outgoing transmission signal. As a current source, the item **112** has a very high impedance (as seen across the secondary winding of transformer **106**), and therefore, only the termination resistor **108** operates to terminate the line (when switched in). Similarly, amplifier **114** is the receive amplifier that generates the receive signal Rx, as is known in the art. Like the current source **112**, the amplifier **114** has an extremely high input impedance and thus does not affect line termination.

Reference is now made to FIG. **6**, which shows a software flow-chart illustrating the operation of the analog front-end element **66** of FIG. **5**. Beginning at step **120**, the element **66** determines whether the local loop *26* is ON-HOOK or OFF-HOOK. As will be appreciated from the foregoing discussion, this decision is made by the controller **80**, which outputs a signal **121** (see FIG. **4**) to the analog front-end **66** indicative of the ON-HOOK/OFF-HOOK status. If the resolution of step **120** is NO, the analog front-end element **66** opens switch **110** (step **122**) to remove the termination resistor **108** from the circuit. That is, if the system detects that a telephone connected to the local loop **26** is OFF-HOOK, it will remove the termination resistor **108** from the circuit, since the line will then be terminated by the OFF-HOOK telephone. Thereafter, operation proceeds to step **122**, wherein data is transmitted in accordance with the band-limited transmission frequency band (e.g., 20 kHz–100 kHz). In accordance with one embodiment of the present invention, the system may emit periodic tones within the audible frequency range to alert a user talking on an attached telephone **30**, **32** (see FIG. **2**) that the local loop **26** is also being used for data transmissions. Thus, a person, for example, speaking in another part of the house over a telephone hearing periodic beeps would know that someone else in the household is using a computer **25** (see FIG. **2**) to communicate data, and therefore, may wish to keep his or her conversation to a minimum, in order to free up the local loop **26**, so that the present invention may obtain a full utilization of the full-banded transmission frequency band, for maximum data throughput.

If the resolution of step **120** is YES, indicating that all telephones **30**, **32** (see FIG. **2**) attached to the local loop **26** are ON-HOOK, then the system ensures that switch **110** is closed thereby placing termination resistor **108** in the circuit, so as to achieve proper line termination (step **130**). Thereafter, the system may transmit data across the local loop utilizing the entire, full-band transmission frequency (i.e., DC to approximately 100 kHz).

Reference is now made to FIG. **7**, which is a software flow-chart illustrating the top-level operation of a system communicating in accordance with the present invention. Beginning at block **140**, the system awaits the initiation of data transmission. This initiation may occur either upon the instruction of a user at the computer **25** (see FIG. **2**), or alternatively, from a remote user that is dialing the phone number of computer **25** to connect up to that computer (this assumes that that computer **25** is in auto answer mode). Once the system has been instructed to begin data communications, it first makes a check (at step **144**) to determine whether the loop is in the OFF-HOOK state. If so, it begins the data communications in the band-limited frequency transition state (step **146**) (e.g., 20 kHz–100 kHz). During the data transmissions, the system will make continuous checks to determine whether the data transmission has ended (step **148**), or whether the band-limiting condition has subsided (step **150**). As previously mentioned, the band-limiting condition is generally identified by the OFF-HOOK detection circuitry. If the end data communications check, at step **148**, resolves to YES, then the system returns to step **140**. If not, the system proceeds to step **150** where it checks for the cessation of the band-limiting condition. If this step resolves to YES, then the system continues the data transmission in the full-band transmission frequency bandwidth (step **154**).

Returning to the decision block **144**, if, upon initiation of data communication, the system determines that all telephones are presently ON-HOOK, then the system proceeds to step **154** where it transmits data in accordance with the full-band data transmission state (i.e., utilizing the full 0 to 100 kHz transmission frequency bandwidth). During transmission in this frequency band, the system periodically checks to see if the data communications has terminated (step **156**), or whether the occurrence of a band-limiting condition has occurred (step **158**). This latter condition occurs, for example, when a person lifts a handset of an attached telephone **30**, **32** (see FIG. **2**). If this occurs, the system proceeds to step **146** and continues the data transmissions in accordance with the band-limited transmission frequency band (20 kHz–100 kHz).

It will be appreciated from a review of the flow-chart of FIG. **7**, that the system, during data transmission, can dynamically shift back and forth between the full-band and band-limited transmission frequency bandwidths as users may lift or reset telephone **30**, **32** (see FIG. **2**) handsets (or as RING conditions occur). It will be appreciated, however, that other band-limiting conditions (other than RING or OFF-HOOK) may be utilized to invoke the frequency shifting feature of the present invention, depending upon the system configuration or other pertinent system factors.

It will be appreciated that the invention described herein could provide a low-cost solution to Internet access for the mass consumer market. In this regard, it could fill the gap between low-cost 33.6 kbps modems and high speed xDSL modems, which require the addition of relatively expensive equipment (such as POTS splitters and POTS filters) at the customer premises, and is labor intensive. The present invention, as described above, generally achieves transmission rates in the range of 64 kbps to 640 kbps.

As described above, the invention utilizes the low frequency portion of the telephone subscriber loop spectrum (roughly DC to approximately 100 kHz) to transport user data. The modulation could be CAP (carrierless amplitude-phase), QAM (quadrature amplitude modulation), DMT (discrete multi-tone), spread spectrum, etc., as the invention is not limited to any particular form. Utilization of the lower frequency portion of the telephone subscriber loop has the advantage of the lowest possible signal attenuation (usually the number one signal impairment in data communications) and low cross-talk. Other advantages are reduced transmission line concerns like reflections due to stubs.

In use, the invention requires a simple bridge (electrical parallel) connection to the subscriber loop or premise wiring. Therefore, one unit would connect (in bridge fashion) at the central office, and one companion unit connect at the customer premises.

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.

We claim:

1 A system for communicating both voice and data between a customer premises and a central office across a communication link comprising:

- means for transmitting data between the customer premises and the central office in a first frequency band, wherein the first frequency band is defined by an upper frequency boundary and a lower frequency boundary;
- means for allocating a second frequency band for transmitting voice information between the customer premises and the central office in the second frequency band;
- means for sensing a band-limiting condition; and
- means for shifting the lower frequency boundary of the first frequency band to a frequency greater than 4 kilohertz in response to the sensed band-limiting condition.

2. The system of claim 1, wherein the means for shifting the lower frequency boundary includes shifting the lower frequency boundary of the first frequency band to at least partially overlap between the first frequency band and the second frequency band, when the band-limiting condition is not present.

3. The system of claim 1, wherein the means for shifting the lower frequency boundary includes shifting the lower frequency boundary of the first frequency band so that there is no overlap between the first frequency band and the second frequency band, when the band-limiting condition is present.

4. The system of claim 1, wherein the means for sensing a band-limiting condition includes sensing an off-hook condition of a telephone electrically connected to the communication link such that the off-hook condition corresponds to the band-limiting condition.

5. The system of claim 1, further including the means for shifting the upper frequency boundary of the first frequency band in response to the sensed band-limiting condition.

6. The system of claim 1, wherein the lower frequency boundary is less than 4 kilohertz.

7. The system of claim 1, wherein the lower frequency boundary is approximately DC.

8. The system of claim 7, wherein the means for shifting the lower frequency boundary shifts the lower frequency boundary upwardly to a frequency of approximately 20 kilohertz in response to the sensed band-limiting condition.

9. The system of claim 1, wherein the means for sensing the band-limiting condition includes the means for detecting the onset of a condition indicative of demand for voice communications.

10. The system of claim 1, wherein the means for sensing the band-limiting condition includes the means for detecting the cessation of a condition indicative of termination of voice communications.

11. The system of claim 10, wherein the means for shifting the lower frequency boundary shifts the lower frequency boundary from a value greater than 4 kilohertz to a value less than 4 kilohertz when the means for detecting detects the cessation of a condition indicative of demand for voice communications.

12. The system of claim 11, further including the means for shifting the upper frequency boundary of the first frequency band when the means for detecting detects the cessation of a condition indicative of demand for voice communications.

13. The system of claim 1, wherein said system is a modem.

14. The system of claim 1, wherein the means for sensing a band-limiting condition includes sensing an impedance change on the communication link such that the sensed impedance change corresponds to the band-limiting condition.

15. The system of claim 1, wherein the means for sensing a band-limiting condition includes sensing a voice conversation on the communication link such that the sensing of the voice conversation corresponds to the band-limiting condition.

16. The system of claim 1, wherein the means for shifting the lower frequency boundary further includes means for decreasing the transmit power of the first frequency band when the band-limiting condition is present.

17. The system of claim 16, wherein the means for shifting the lower frequency boundary further includes means for increasing the transmit power of the first frequency band when the band-limiting condition is not present.

18. The system of claim 1, wherein the means for shifting the lower frequency boundary further includes means for increasing the transmit power of the first frequency band when the band-limiting condition is present.

19. The system of claim 18, wherein the means for shifting the lower frequency boundary further includes means for decreasing the transmit power of the first frequency band when the band-limiting condition is not present.

20. The system of claim 1, further including a means for emitting an audible signal when the first frequency band is present and when the band-limiting condition is present in order to indicate communication of the data.

21. The system of claim 1, further comprising a means to allocate a first power spectral density (PSD) when the band-limiting condition is present and a means to allocate a second PSD when the band-limiting condition is not present.

22. The system of claim 1, further comprising a front end element, said front end element comprising:

- a line termination resistor; and
- means for switching disposed in series with the line termination resistor, wherein the line termination resistor and the means for switching are disposed in parallel to a pair of conductors residing in the communication link such that the means for switching connects the line termination resistor between said pair of conductors when the band-limiting condition is not present, and such that the means for switching disconnects the line termination resistor from said pair of conductors when the band-limiting condition is present.

23. The system of claim 1, wherein the lower frequency boundary of the first frequency band is initially less than 4 kilohertz.

24. The system of claim 1, wherein the lower frequency boundary of the first frequency band is initially greater than 4 kilohertz.

25. The system of claim 1, wherein the means for shifting further comprises means for dynamically shifting.

26. A method for communicating both voice and data between a customer premises and a central office across a communication link comprising the steps of:

transmitting data between the customer premises and the central office in a first frequency band, wherein the first frequency band is defined by an upper frequency boundary and a lower frequency boundary;

allocating a second frequency band for transmitting voice information between the customer premises and the central office in the second frequency band;

sensing a band-limiting condition; and

shifting the lower frequency boundary of the first frequency band to a frequency greater than 4 kilohertz in response to the sensed band-limiting condition.

27. The method of claim **26**, wherein the step of shifting the lower frequency boundary includes shifting the lower frequency boundary of the first frequency band to at least partially overlap between the first frequency band and the second frequency band, when the band-limiting condition is not present.

28. The method of claim **26**, wherein the step of shifting the lower frequency boundary includes shifting the lower frequency boundary of the first frequency band so that there is no overlap between the first frequency band and the second frequency band, when the band-limiting condition is present.

29. The method of claim **26**, wherein the step of sensing a band-limiting condition includes sensing an off-hook condition of a telephone electrically connected to the communication link.

30. The method of claim **26**, further including the step of shifting the upper frequency boundary of the first frequency band in response to the sensed band-limiting condition.

31. The method of claim **26**, wherein the lower frequency boundary is less than 4 kilohertz.

32. The method of claim **26**, wherein the lower frequency boundary is approximately DC.

33. The method of claim **32**, wherein the step of shifting the lower frequency boundary includes the step of shifting the lower frequency boundary upwardly to a frequency of approximately 20 kilohertz in response to the sensed band-limiting condition.

34. The method of claim **26**, wherein the step of sensing the band-limiting condition includes the step of detecting the onset of a condition indicative of demand for voice communications.

35. The method of claim **26**, wherein the step of sensing the band-limiting condition includes the step of detecting the cessation of a condition indicative of the termination of voice communications.

36. The method of claim **35**, wherein the step of shifting the lower frequency boundary includes the step of shifting the lower frequency boundary from a value greater than 4 kilohertz to a value less than 4 kilohertz when the step of detecting detects the cessation of a condition indicative of demand for voice communications.

37. The method of claim **36**, further including the step of shifting the upper frequency boundary of the first frequency band when the means for detecting detects the cessation of a condition indicative of demand for voice communications.

38. The method of claim **35**, wherein said system is a modem.

39. The method of claim **35**, wherein the step of shifting the lower frequency boundary includes the step of shifting the lower frequency boundary from a value greater than 4 kilohertz to a lower value, the lower value greater than 4 kilohertz, when the step of detecting detects the cessation of a condition indicative of demand for voice communications.

40. The method of claim **26**, wherein the step of sensing a band-limiting condition includes sensing an impedance change on the communication link such that the sensed impedance change corresponds to the band-limiting condition.

41. The method of claim **26**, wherein the step of sensing a band-limiting condition includes sensing a voice conversation on the communication link such that the sensing of the voice conversation corresponds to the band-limiting condition.

42. The method of claim **26**, wherein the step of shifting the lower frequency boundary further includes the step of increasing the transmit power of the first frequency band when the band-limiting condition is not present.

43. The method of claim **42**, wherein the step of shifting the lower frequency boundary further includes the step of decreasing the transmit power of the first frequency band when the band-limiting condition is present.

44. The method of claim **26**, wherein the step of shifting the lower frequency boundary further includes the step of decreasing the transmit power of the first frequency band when the band-limiting condition is not present.

45. The method of claim **44**, wherein the step of shifting the lower frequency boundary further includes the step of increasing the transmit power of the first frequency band when the band-limiting condition is present.

46. The system of claim **1**, wherein the means for shifting the lower frequency boundary further includes means for asymmetrically shaping a portion of the first frequency band when the band-limiting condition is present.

47. The method of claim **26**, wherein the step of shifting the lower frequency boundary further includes the step of asymmetrically shaping a portion of the first frequency band when the band-limiting condition is present.

48. The method of claim **26**, further including the step of emitting an audible signal when the first frequency band is present and when the band-limiting condition is present in order to indicate communication of the data.

49. The method of claim **26**, further comprising the step of allocating a first power spectral density (PSD) when the band-limiting condition is present and the step of allocating a second PSD when the band-limiting condition is not present.

50. The method of claim **26**, further comprising the steps of:

switching a line termination resistor so that the line termination resistor is connected in parallel with a pair of conductors residing in the communication link when the step of sensing senses the band-limiting condition; and

switching the line termination resistor so that the line termination resistor is disconnected from the pair of conductors residing in the communication link when the step of sensing senses an absence of the band-limiting condition.

51. The method of claim **26**, wherein the step of shifting further comprises dynamically shifting.

52. The method of claim **26**, wherein the step of shifting further comprises shifting the lower frequency boundary of the first frequency band from an initial frequency less than 4 kilohertz to the frequency greater than 4 kilohertz in response to the sensed band-limiting condition.

53. The method of claim **26**, wherein the step of shifting further comprises shifting the lower frequency boundary of the first frequency band from an initial frequency greater than 4 kilohertz to a greater frequency in response to the sensed band-limiting condition.

54. A system for communicating both voice and data between a customer premises and a central office across a communication link comprising:

means for transmitting data between the customer premises and the central office in a first frequency band, wherein the first frequency band is defined by an upper frequency boundary and a lower frequency boundary;

means for sensing a band-limiting condition;

means for shifting the lower frequency boundary of the first frequency band to a frequency greater than 4 kilohertz in response to the sensed band-limiting condition; and

means for emitting an audible signal when the first frequency band is present and when the band-limiting condition is present in order to indicate communication of the data.

55. The system of claim **54**, wherein the means for shifting further comprises means for dynamically shifting.

56. The system of claim **54**, further comprising means for shifting the lower frequency boundary of the first frequency band from an initial frequency less than 4 kilohertz to the frequency greater than 4 kilohertz in response to the sensed band-limiting condition.

57. The system of claim **54**, further comprising means for shifting the lower frequency boundary of the first frequency band from an initial frequency greater than 4 kilohertz to a greater frequency in response to the sensed band-limiting condition.

58. A method for communicating both voice and data between a customer premises and a central office across a communication link comprising the steps of:

transmitting data between the customer premises and the central office in a first frequency band, wherein the first frequency band is defined by an upper frequency boundary and a lower frequency boundary;

sensing a band-limiting condition;

shifting the lower frequency boundary of the first frequency band to a frequency greater than 4 kilohertz in response to the sensed band-limiting condition; and

emitting an audible signal when the first frequency band is present and when the band-limiting condition is present in order to indicate communication of the data.

59. The method of claim **58**, wherein the step of shifting further comprises dynamically shifting.

60. The method of claim **58**, wherein the step of shifting further comprises shifting the lower frequency boundary of the first frequency band from an initial frequency less than 4 kilohertz to the frequency greater than 4 kilohertz in response to the sensed band-limiting condition.

61. The method of claim **58**, wherein the step of shifting further comprises shifting the lower frequency boundary of the first frequency band from an initial frequency greater than 4 kilohertz to a greater frequency in response to the sensed band-limiting condition.

62. A system for communicating both voice and data between a customer premises and a central office across a communication link comprising:

means for allocating a first frequency band for transmitting data between the customer premises and the central office, wherein the first frequency band is defined by an upper frequency boundary and a lower frequency boundary;

means for determining the initiation of data transmission on the first frequency band;

means for sensing a band-limiting condition;

means for setting the lower frequency boundary of the first frequency band: to a first lower frequency greater than 4 kilohertz in response to determining initiation of data transmission and the sensing of the band-limiting condition; and

means for setting the lower frequency boundary of the first frequency band to a second lower frequency, the second lower frequency less than the first lower frequency, in response to determining initiation of data transmission and to the sensing of an absence of the band-limiting condition.

63. The system of claim **62**, wherein the means for sensing the band-limiting condition further includes means for sensing transmitted voice information between the customer premises and the central office in a second frequency band, wherein the sensing of voice information corresponds to the band-limiting condition.

64. The system of claim **62**, wherein means for determining the initiation of data transmission further includes means for sensing an instruction from the user of a computer.

65. The system of claim **62**, wherein means for determining the initiation of data transmission further includes means for sensing dialing of a phone number of a computer operating in an auto answer mode.

66. A method for communicating both voice and data between a customer premises and a central office across a communication link comprising:

allocating a first frequency band for transmitting data between the customer premises and the central office, wherein the first frequency band is defined by an upper frequency boundary and a lower frequency boundary;

determining the initiation of data transmission on the first frequency band;

sensing a band-limiting condition;

setting the lower frequency boundary of the first frequency band to a first lower frequency greater than 4 kilohertz in response to determining initiation of data transmission and the sensing of the band-limiting condition; and

setting the lower frequency boundary of the first frequency band to a second lower frequency, the second lower frequency less than the first lower frequency, in response to determining initiation of data transmission and to the sensing of an absence of the band-limiting condition.

67. The method of claim **66**, wherein the step of sensing the band-limiting condition further includes the step of sensing transmitted voice information between the customer premises and the central office in a second frequency band, wherein the sensing of voice information corresponds to the band-limiting condition.

68. The method of claim **66**, wherein the step of determining the initiation of data transmission further includes the step of sensing an instruction from the user of a computer.

69. The method of claim **66**, wherein the step of determining the initiation of data transmission further includes the step of sensing dialing of a phone number of a computer operating in an auto answer mode.

70. The method of claim **66**, wherein step of setting the lower frequency boundary of the first frequency band to the first lower frequency further comprises setting the first lower frequency to an initial frequency less than 4 kilohertz.

71. The method of claim **66**, wherein the step of setting the lower frequency boundary of the first frequency band to the first lower frequency further comprises setting the first lower frequency to an initial frequency greater than 4 kilohertz.

72. The method of claim **66**, wherein the step of setting the lower frequency boundary of the first frequency band to

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the second lower frequency further comprises setting the second lower frequency to less than 4 kilohertz.

73. The method of claim 66, wherein the step of setting the lower frequency boundary of the first frequency band to the second lower frequency further comprises setting the second lower frequency to greater than 4 kilohertz.

74. A system for communicating both voice and data between a customer premises and a central office across a communication link comprising:

means for sensing a band-limiting condition;

a line termination resistor; and

means for switching disposed in series with the line termination resistor, wherein the line termination resistor is disposed in parallel to a pair of conductors residing in the communication link such that the means for switching connects the line termination resistor between said pair of conductors when the band-limiting condition is not present, and such that the means for switching disconnects the line termination resistor from said pair of conductors when the band-limiting condition is present.

75. A method for communicating both voice and data between a customer premises and a central office across a communication link comprising the steps of:

sensing a band-limiting condition;

switching a line termination resistor such that the line termination resistor is connected in parallel with a pair of conductors residing in the communication link when the step of sensing senses the band-limiting condition; and

switching a line termination resistor such that the line termination resistor is disconnected from the pair of conductors residing in the communication link when the step of sensing senses an absence of the band-limiting condition.

76. A computer readable medium having a program for communication both voice and data over a communication link, the program comprising logic configured to perform the steps of:

transmitting data between the customer premises and the central office in a first frequency band, wherein the first frequency band is defined by an upper frequency boundary and a lower frequency boundary;

allocating a second frequency band for transmitting voice information between the customer premises and the central office in the second frequency band;

sensing a band-limiting condition;

shifting the lower frequency boundary of the first frequency band to a first frequency to a frequency greater than 4 kilohertz when the step of sensing senses the band-limiting condition, and

shifting the lower frequency boundary of the first frequency band to a second lower frequency in response to determining initiation of data transmission and to the sensing of an absence of the band-limiting condition.

77. The computer readable medium of claim 76, further having logic configured to perform the step of shifting the lower frequency boundary of the first frequency band to a second frequency when the step of sensing senses an absence of the band-limiting condition.

78. The computer readable medium of claim 76, further having logic configured to perform the step of switching a line termination resistor such that the line termination resistor is connected in parallel with a pair of conductors residing in the communication link when the step of sensing senses the band-limiting condition.

79. The computer readable medium of claim 78, further having logic configured to perform the step of switching the line termination resistor such that the line termination resistor is disconnected from the pair of conductors residing in the communication link when the step of sensing senses an absence of the band-limiting condition.

80. The computer readable medium of claim 76, wherein the logic configured to shift the lower frequency boundary of the first frequency band to the first frequency shifts the first frequency from an initial frequency greater than 4 kilohertz.

81. The computer readable medium of claim 76, wherein the logic configured to shift the lower frequency boundary of the first frequency band to the first frequency shifts the first frequency from an initial frequency less than 4 kilohertz.

82. The computer readable medium of claim 76, wherein the logic configured to shift the lower frequency boundary of the first frequency band to the second lower frequency shifts the second lower frequency to greater than 4 kilohertz.

83. The computer readable medium of claim 76, wherein the logic configured to shift the lower frequency boundary of the first frequency band to the second lower frequency shifts the second lower frequency to less than 4 kilohertz.

* * * * *

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US006154524A

United States Patent [19]

Bremer

[11] **Patent Number: 6,154,524**

[45] **Date of Patent: *Nov. 28, 2000**

[54] **METHOD AND APPARATUS FOR AUTOMATICALLY AND ADAPTIVELY ADJUSTING TELEPHONE AUDIO QUALITY AND DSL DATA RATE IN A DSL SYSTEM**

[75] Inventor: **Gordon Bremer**, Clearwater, Fla.

[73] Assignee: **Paradyne Corporation**, Largo, Fla.

[*] Notice: This patent is subject to a terminal disclaimer.

[21] Appl. No.: **09/240,465**

[22] Filed: **Jan. 29, 1999**

[51] **Int. Cl.**[7] **H04M 1/24**

[52] **U.S. Cl.** **379/27**; 379/1; 379/12; 379/24; 379/28

[58] **Field of Search** 379/1–2, 26–29, 379/32, 34; 375/222, 224, 225, 228

[56] **References Cited**

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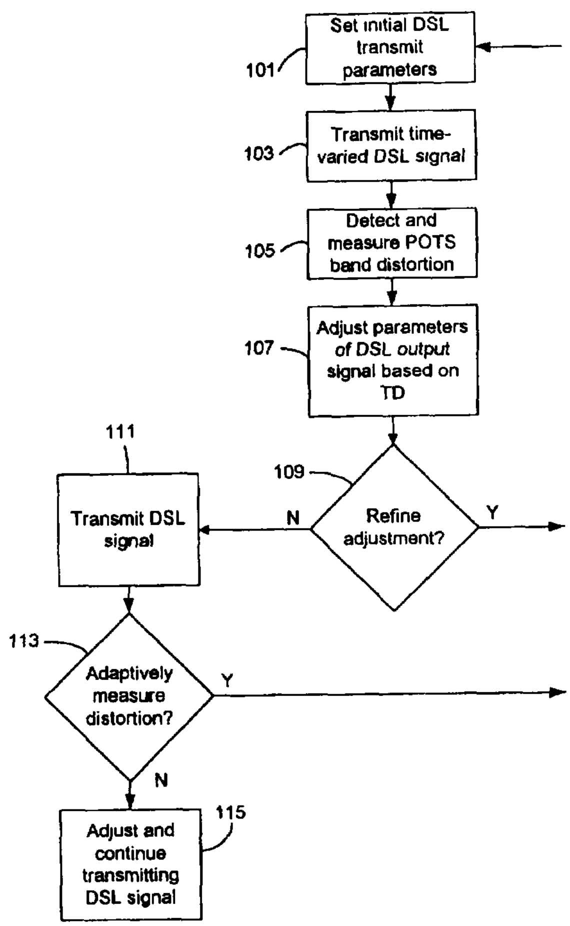
6,014,425 1/2000 Bingel et al. 379/27

Primary Examiner—Stella Woo
Assistant Examiner—Binh K. Tieu
Attorney, Agent, or Firm—Thomas, Kayden, Horstemeyer & Risley, L.L.P.

[57] **ABSTRACT**

In a communications environment where it is desirable to allow the simultaneous transmission of digital subscriber line (DSL) signals and conventional plain old telephone service (POTS) signals on a single two-wire communication line, a method and apparatus is configured to automatically measure the distortion caused by a telephone or other attached device such as a dial modem due to the presence of a DSL signal on the same communication line, and to use such measurement to automatically and adaptively adjust the DSL output signal to achieve the highest DSL data rate commensurate with an acceptable level of distortion as required by the POTS devices.

19 Claims, 9 Drawing Sheets



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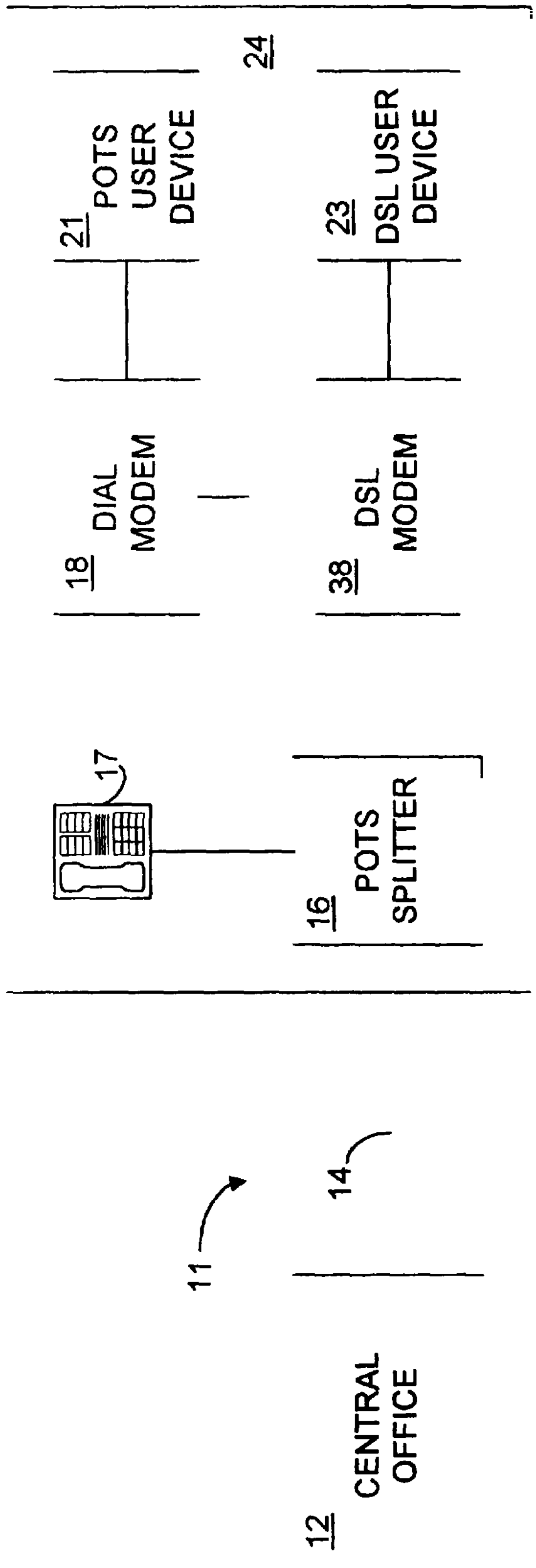

Fig. 1
(PRIOR ART)

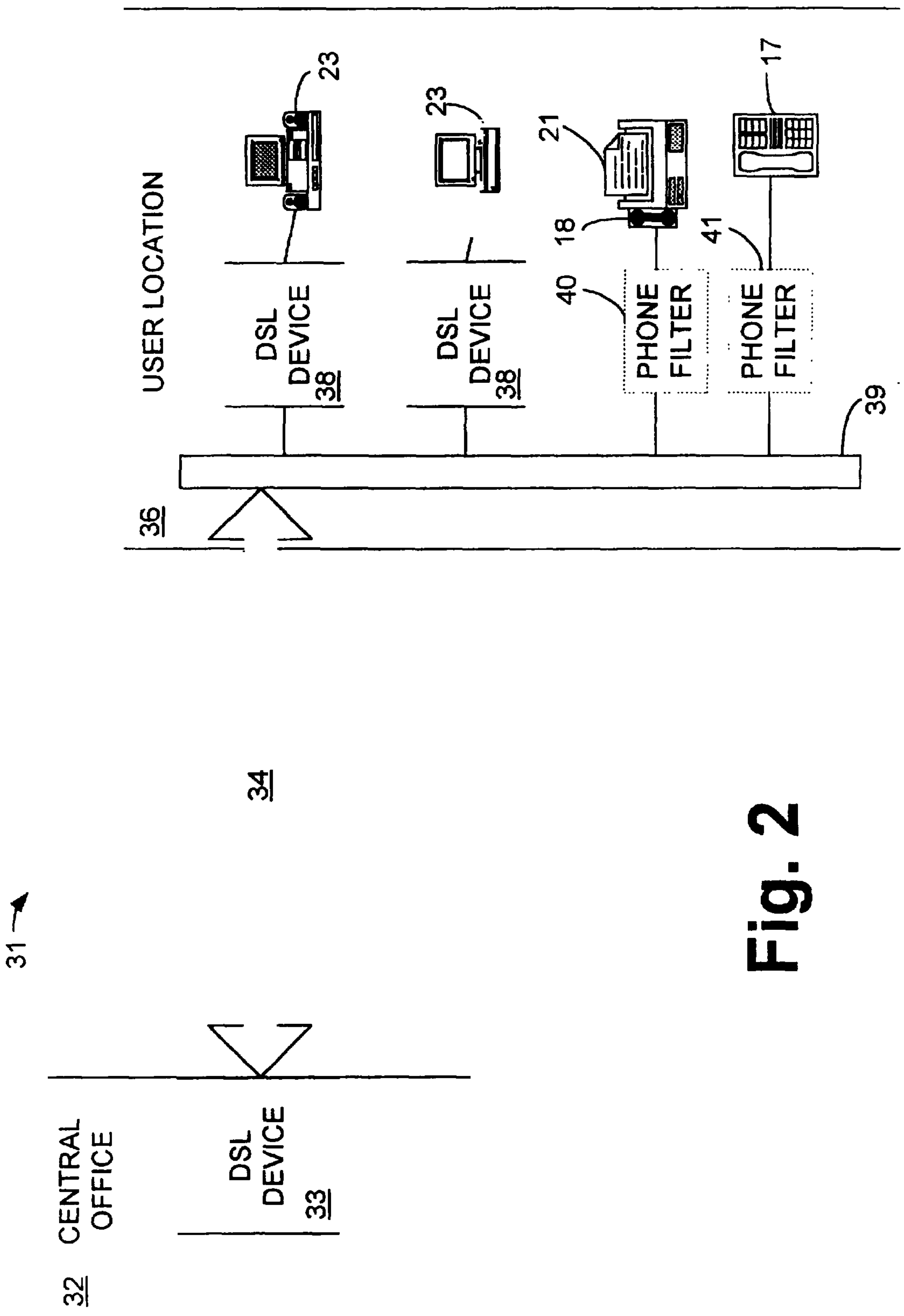
USER LOCATION
36
DSL DEVICE 38
DSL DEVICE 38
PHONE FILTER
PHONE FILTER
23
23
21
18
17
40
41
39
34
31
CENTRAL OFFICE
32
DSL DEVICE 33

Fig. 2

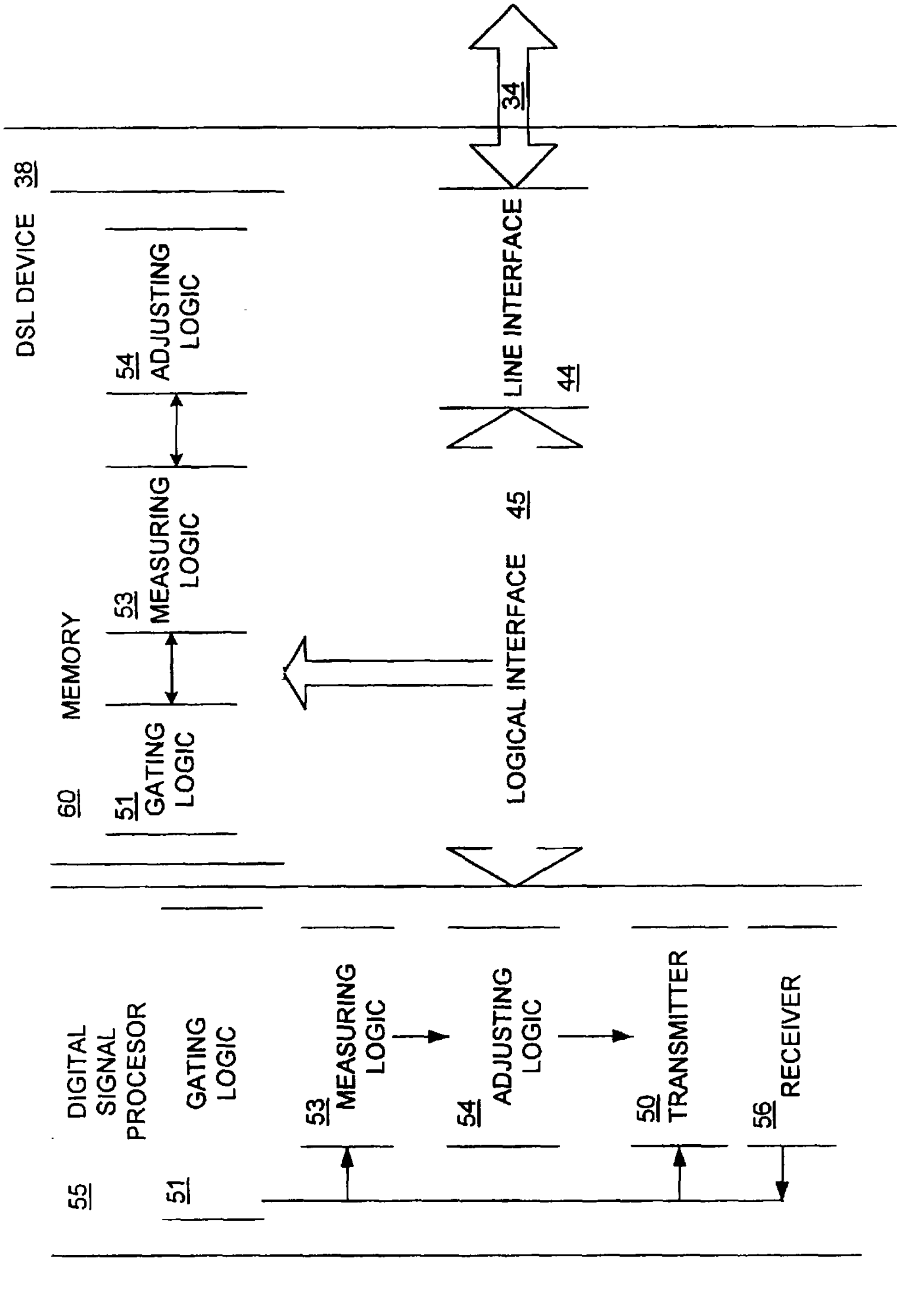
34
DSL DEVICE 38
54 ADJUSTING LOGIC
53 MEASURING LOGIC
MEMORY
60
51 GATING LOGIC
LINE INTERFACE 44
LOGICAL INTERFACE 45
DIGITAL SIGNAL PROCESOR
55
GATING LOGIC
51
53 MEASURING LOGIC
54 ADJUSTING LOGIC
50 TRANSMITTER
56 RECEIVER

Fig. 3

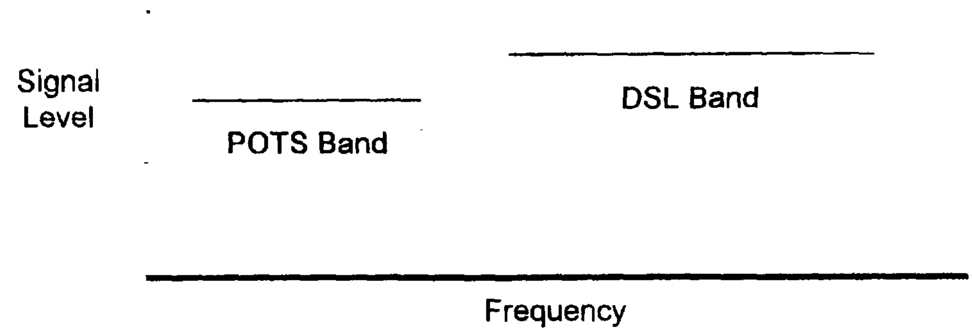

FIG. 4A

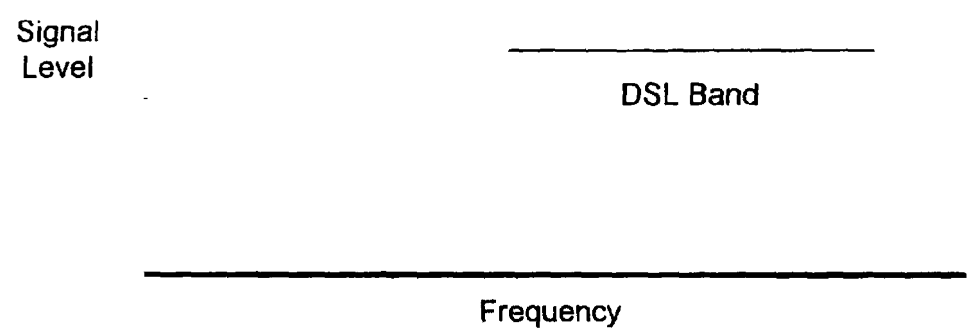

FIG. 4B

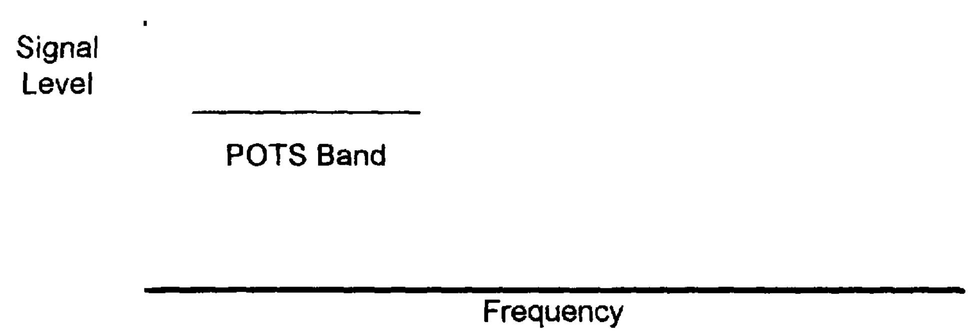

FIG. 4C

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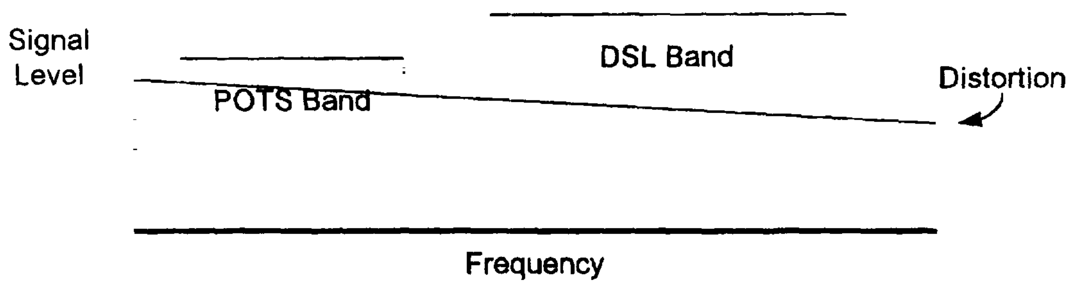

FIG. 5

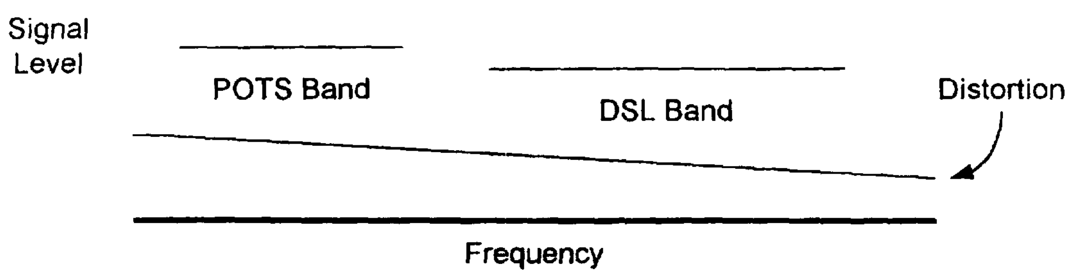

FIG. 6

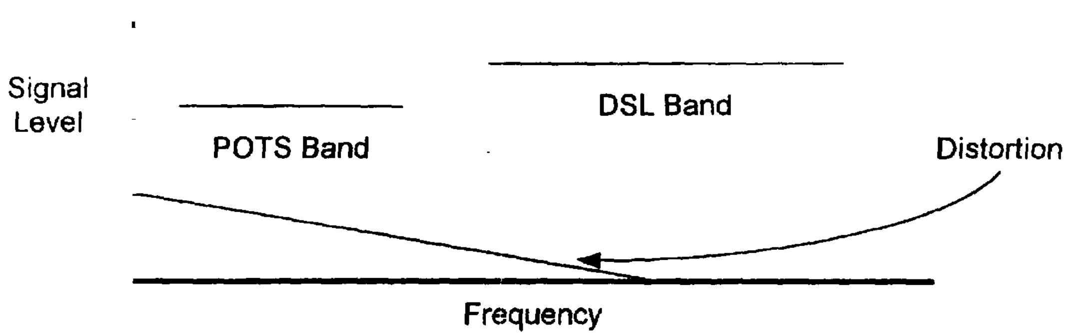

FIG. 7

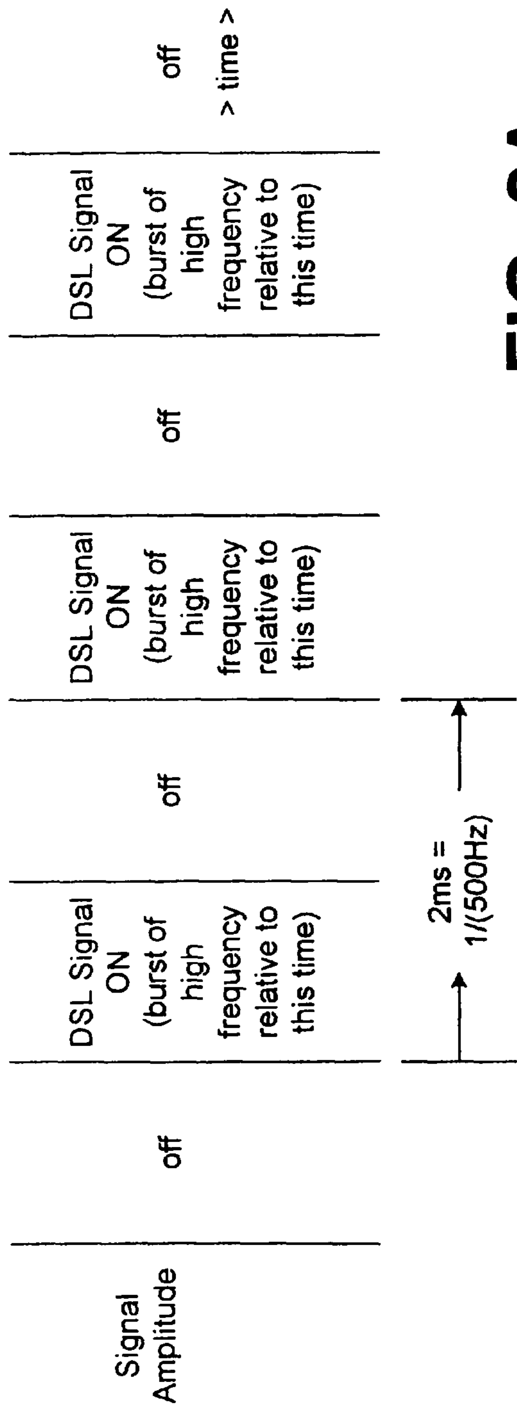

FIG. 8A

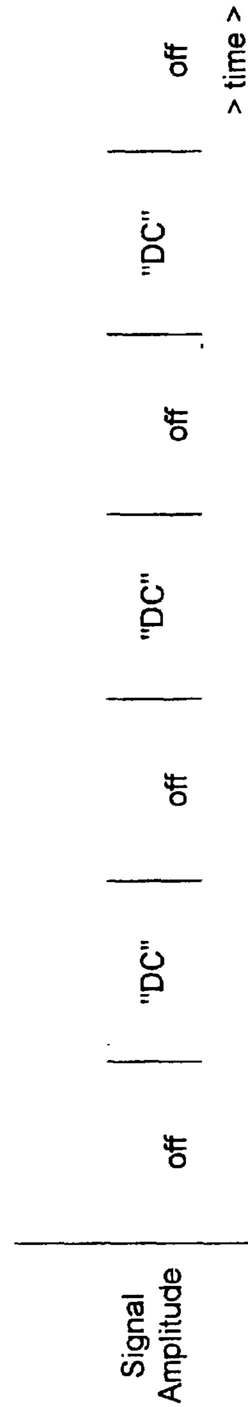

FIG. 8B

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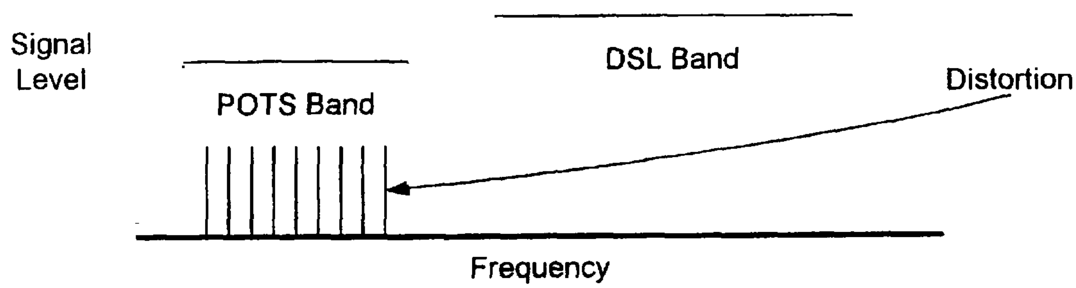
Signal
Level
POTS Band
DSL Band
Distortion
Frequency

FIG. 9

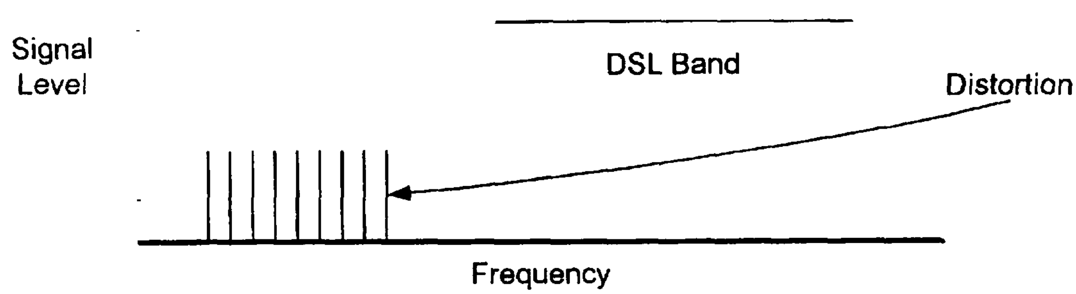
Signal
Level
DSL Band
Distortion
Frequency

FIG. 10

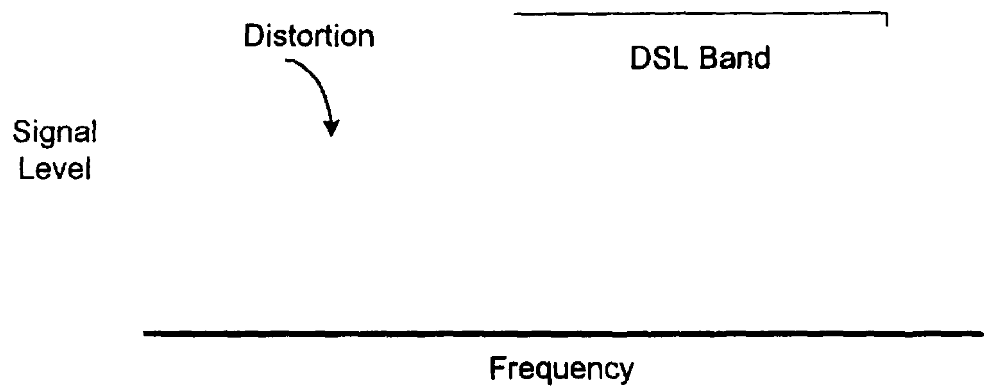

FIG. 11A

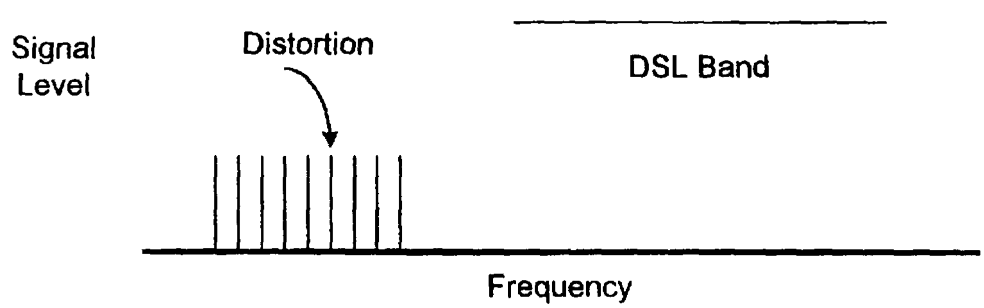

FIG. 11B

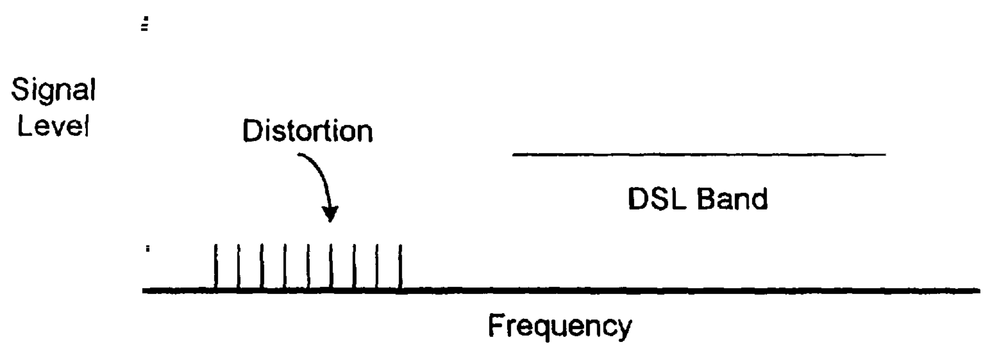

FIG. 11C

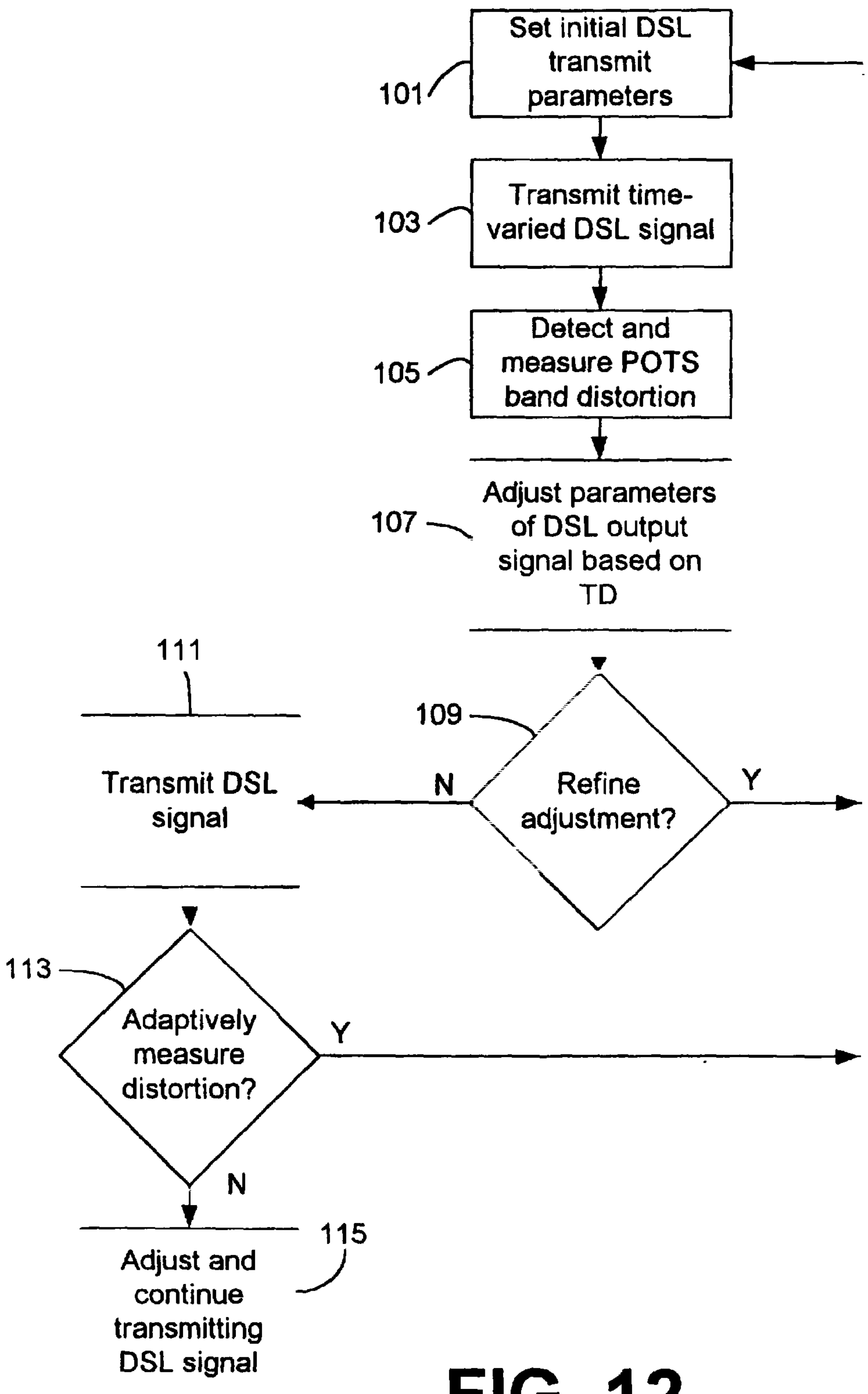
Set initial DSL transmit parameters
101
Transmit time-varied DSL signal
103
Detect and measure POTS band distortion
105
Adjust parameters of DSL output signal based on TD
107
111
109
Transmit DSL signal
N
Refine adjustment?
Y
113
Adaptively measure distortion?
Y
N
115
Adjust and continue transmitting DSL signal

FIG. 12

METHOD AND APPARATUS FOR AUTOMATICALLY AND ADAPTIVELY ADJUSTING TELEPHONE AUDIO QUALITY AND DSL DATA RATE IN A DSL SYSTEM

CROSS REFERENCE TO RELATED APPLICATIONS

This application claims priority to and the benefit of the filing date of co-pending and commonly assigned provisional application entitled AUTOMATIC AND ADAPTIVE TELEPHONE AUDIO QUALITY ADJUSTMENT AND DATA RATE CHANGES IN A DSL SYSTEM, assigned Serial No 60/072,826, filed Jan. 28, 1998, and hereby incorporated by reference. The present application is also related to copending application entitled METHOD AND APPARATUS FOR AUTOMATICALLY DETECTING AND MEASURING DISTORTION IN A DSL SYSTEM, filed on even date herewith, under express mail no. EL068409195US

TECHNICAL FIELD

The present invention relates generally to communication devices, and more particularly, to a method and apparatus for automatically and adaptively adjusting the parameters of a DSL transmit signal in a communications system that includes digital subscriber line (DSL) signals and conventional telecommunications signals to maximize the data rate while keeping the distortion within acceptable levels.

BACKGROUND OF THE INVENTION

In the field of data communications, modems are used to convey information from one location to another. Digital subscriber line (DSL) technology now enables DSL devices, such as DSL modems, to communicate large amounts of digital data. Typically in a communications environment, plain old telephone service (POTS) type devices (such as telephones, facsimile machines, and dial modems) conveying conventional telecommunications signals are connected to the same subscriber wire pair as DSL devices at the user's location, which is typically remote from the telephone company's central office location, via a wire pair provided by the local telephone company.

Because passband DSL signals, such as asymmetric digital subscriber line (ADSL) and rate adaptive digital subscriber line (RADSL) modem signals, typically occupy only the frequency band above the audio band, these DSL signals have traditionally been isolated from all POTS type devices (such as telephones or dial modems) by a splitter or filter system installed at the user (remote) location. Such a splitter is typically known in the field of telephony communications as a POTS splitter. The POTS splitter typically serves two purposes: (1) it attenuates the DSL signals so that they do not significantly appear at the input of the POTS devices, and (2) it attenuates the POTS so that they do not significantly appear at the input of the DSL devices. In particular, the POTS filter attempts to attenuate DSL signals appearing at the input of the POTS devices in the audio band to an inaudible level, and also attempts to attenuate DSL signals above the audio band to a level low enough so that distortion inside the POTS type devices does not adversely affect their performance.

Although POTS splitters have been effective, such splitters are undesirable for many applications because of installation and cost issues. Recognizing the undesirable attributes of POTS splitters, efforts to eliminate them have begun in the DSL industry and standards bodies. These attempts call for "splitter-less-DSL" systems wherein the DSL devices and the POTS devices are all directly connected to the two-wire communications channel without the use of a POTS splitter.

In splitter-less operation, it is necessary for the DSL device to: (1) filter its output signal to ensure that signals in the audio band are below an audible level, and (2) reduce its output (transmit signal power to a level that does not cause adverse distortion in the POTS devices. The reduction in output signal power is a crucial aspect of splitter-less operation. Unfortunately, such reduction may dramatically reduce data performance, in some cases to an unacceptable level. Due to line losses, reduction in the output signal power level also reduces the reach of the subscriber loop

To facilitate production of a splitter-less DSL device, such as a DSL modem, that can be installed in a variety of systems without field adjustment, the DSL device transmit level must be pre-set to the lowest level commensurate with the worst distortion situation expected for the potential universe of attached POTS devices. As noted above, such a reduced output level may be undesirable in many installations. Although many installations might tolerate a higher output level, this level could only be determined by trial and error for each installation, which is a cumbersome process and impractical for mass deployment of DSL splitter-less devices for consumer applications.

Due to the problems involved in using many DSL devices in splitter-less operation, it has been proposed that phone filters (also known as distributed POTS splitters) be placed at the interface to each item of POTS type equipment at the remote location. A phone filter is a bi-directional lowpass filter that attenuates as much of the DSL modem signal as practical from appearing at the interface to the POTS type devices, such as telephones and dial modems. A phone filter can markedly improve splitter-less operation by permitting the transmit power level of the DSL device to be increased above the level that would produce distortion in splitter-less operation. However, phone filters also have cost and installation drawbacks and they are less suitable for some types of DSL operation than a system utilizing a POTS splitter. Thus, it is desirable to eliminate phone filters where they are not required, for example on a particular POTS device that is immune to distortion.

Therefore, there is a need in the industry for a method and apparatus for automatically and adaptively adjusting the parameters of the transmit signal of a DSL device to maximize the data rate of the DSL signal being transmitted while keeping the distorting effects of the DSL signal on the POTS devices to an acceptable level

SUMMARY OF THE INVENTION

A technique is presented that permits a DSL-type device on a splitter-less wire-pair communication channel to automatically and adaptively adjust parameters of its transmit signal, including the transmit power level, to exchange data rate performance for audio performance of a telephone or data performance of an attached POTS device such as a dial modem on that same wire pair that, in the presence of a DSL-type line signal, is causing distortion either: (1) detrimental to the data performance of the DSL-type device; (2) likely to be detrimental to the audio quality of the telephone and telephone system including the introduction of audio noise; or (3) detrimental to the data rate performance of the dial modem.

Absent such parameter adaptation, the data rate of the DSL-type device must be set at the lowest data rate com-

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mensurate with the worst distortion situation expected for the universe of potential attached telephones and modems, with or without phone filters.

The technique utilizes the fact that certain envelope-modulated or time-gated DSL-type passband signals cause certain predicable audio frequency band signals on the wire pair connected to a distorting POTS-type device, such as a telephone or dial modem, and that these audio frequency band signals can be detected and quantified in the DSL-type device, even in the presence of other audio signals that ordinarily occur on the wire pair channel. Given continuing quantification, an acceptable distortion level for the system can be determined and the transmit parameters of the DSL device can be adaptively changed to keep the distortion from exceeding the acceptable level while maximizing the DSL data rate.

The technique is implemented by varying the output signal of a DSL-type device on a wire pair communication channel according to a time sequence to produce a time-varied DSL signal, measuring the non-linear distortion on the communication channel in the presence of the time-varied DSL signal, adjusting the transmit parameters of the output signal based on the measured distortion, and transmitting the adjusted output signal.

BRIEF DESCRIPTION OF THE SEVERAL VIEWS OF THE DRAWINGS

The present invention, as defined in the claims, can be better understood with reference to the following drawings. The drawings are not necessarily to scale, emphasis instead being placed on clearly illustrating the principles of the present invention.

FIG. 1 is a schematic view illustrating at prior art communications environment in which a POTS splitter is employed to isolate a DSL device from conventional POTS-type devices;

FIG. 2 is a schematic view of a multipoint communications environment including DSL devices employing the automatic output parameter adjustment method and apparatus of the present invention;

FIG. 3 is a simplified schematic view of a DSL device of FIG. 2 employing the automatic output parameter adjustment method and apparatus of the present invention;

FIGS. 4A–4C are simplified illustrations of the signal level and frequency placement of POTS band signals and DSL band signals on the two-wire communications channel of FIG 1;

FIG. 5 is a simplified illustration of the signal level and frequency placement of POTS band signals, DSL band signals, and distortion on the two-wire communications channel of FIG. 2, in which no phone filter is used;

FIG. 6 is a simplified illustration of the signals of FIG. 5 measured at the conventional telephone interface, where a phone filter has been used to reduce the level of the DSL signal, and hence the distortion.

FIG. 7 is a simplified illustration of the signals of FIG. 5 measured at the DSL device, where a phone filter has been used to reduce the level of the DSL signal, and hence the distortion;

FIGS. 8A and 8B are simplified illustrations of the correlation of the DSL signal and the distortion components of that, signal when the DSL signal is time-varied according to the present invention;

FIG. 9 is a simplified illustration of the signal level and frequency placement of POTS band signals, DSL band signals, and distortion on the two-wire communications channel of FIG. 2 when the DSL signal is time-varied according to the present invention;

FIG. 10 is a further simplification of FIG. 9, in which the POTS band designation has been eliminated;

FIGS. 11A–11C are simplified illustrations of three DSL device transmit signal levels and the corresponding distortion components; and

FIG. 12 is a flow chart illustrating the operation of the method and apparatus for automatically and adaptively adjusting the parameters of the output signal of a DSL device to maximize the DSL data rate while keeping the distortion within acceptable levels.

DETAILED DESCRIPTION OF THE INVENTION

The present invention can be implemented in software, hardware, firmware, or a combination thereof. In the preferred embodiment, the elements of the present invention are implemented in software that is stored in a memory and that configures and drives a suitable digital signal processor (DSP) situated in the respective DSL-type device. However, the foregoing software can be stored on any computer-readable medium for use by or in connection with any suitable computer-related system or method. In the context of this document, a computer-readable medium is 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 (CD-ROM) (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.

FIG. 1 is a schematic view illustrating a prior art communications environment **11**. Communications channel **14** is a conventional two-wire communications system that typically connects a telephone company central office location **12** to a remote user location **24**. Remote user location **24** is typically a residential or business location and includes POTS devices such as telephone **17**, dial modem **18**, and user device **21**, as well as DSL devices, such as DSL modem **38** and user device **23**. A POTS splitter **16** is employed to connect the telephone **17**, dial modem **18**, and DSL modem **38** to the communications channel **14**. POTS splitter **16** is required in this application in order to isolate the telephone **17** and dial modem **18** from the DSL modem **38**. POTS splitter **16** is typically located at remote location **24**.

The POTS splitter **16** typically serves two purposes: (1) it attenuates the DSL signals so that they do not significantly appear at the input of the telephone **17** or dial modem **18**, and (2) it attenuates the telephone **17** or dial modem **18** transmit signals so that they do not significantly appear at the input of the DSL modem **38**. In particular, the POTS splitter **16** attempts to attenuate the DSL modem **38** signals appearing at the telephone **17** or dial modem **18** input in the audio band to an inaudible level, and also attempts to attenuate

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DSL modem 38 signals above the audio band to a level low enough so that distortion in the telephone 17 or dial modem 18 does not adversely affect their performance.

FIG. 2 is a schematic view of a multipoint communications environment 31 in which POTS devices 17 and 18, and DSL devices 33 and 38 operate. In this figure, the DSL devices 38 employ the automatic output parameter adjustment method and apparatus of the present invention. Notably, communications environment 31 does not include a POTS splitter. Optional phone filters 40 and 41 may be used to reduce distortion, as discussed hereinafter.

User location 36 is connected to central office location 32 via communication channel 34. DSL device 33 is located at central office location 32, while at least one DSL device 38 is located at the user location 36, which is remote from central office location 32. Communication channel 34 is typically the two-wire communication channel that extends between a telephone company central office and a remote residential, business, or any other location served by local telephone service. Remote location 36 may contain a plurality of remote DSL devices 38 connecting a plurality of user devices 23 to communication channel 34 via communication bus 39. Communication bus 39 is illustratively the wiring infrastructure used throughout a remote location to connect remote DSL devices 38 to communication channel 34. In addition, remote location 36 may contain conventional POTS devices 17 and 18. Illustratively, conventional telephone 17 and dial modem 18 (which is not shown in FIG. 2, but is contained within user device 21, which is illustratively a facsimile machine) are connected to communication bus 39, and thus to communication channel 34. By using DSL devices 38 employing the concepts and features of the automatic output parameter adjustment method and apparatus of the present invention, it is possible to connect POTS devices 17 and 18 directly to communication channel 34 without experiencing an unacceptable level of distortion. Optionally, phone filters 40 and 41 may be used to allow the output of DSL devices 38 to be increased further while maintaining the distortion at POTS devices 17 and 18 within acceptable levels.

Now referring to FIG. 3, shown is a schematic view illustrating DSL device 38 of FIG. 2, including the automatic output parameter adjustment logic 54 of the present invention. Typically, DSL device 38 will transmit signals to the central office 32 of FIG. 2 over communication channel 34. Similarly, central office 32 will transmit signals to DSL device 38. DSL device 38 contains conventional components as are known in the art of data communications. Digital signal processor (DSP) 55 controls the operation of and includes transmitter 50 and receiver 56 of DSL device 38. DSP 55 couples through logical interface 45 to line interface 44 to gain access to communication channel 34. Also included in DSP 55 of DSL device 38 are gating logic 51, distortion detection and measuring logic 53, and output adjusting logic 54, which enable DSL device 38 to perform the automatic and adaptive adjustment functions of the present invention, as discussed hereinbelow. Also contained within DSL device 38 is memory 60, which also includes gating logic 51, distortion detection and measuring logic 53, and output adjusting logic 54. In a preferred embodiment, the logic of the present invention is executed within DSP 55 and is therefore shown as residing in both memory 60 and DSP 55.

Still referring to FIG. 3, the output (transmit) DSL signal of DSL device 38 is passed to communication channel 34 (via logical interface 45 and line interface 44) from transmitter 50. Transmitter 50 is controlled by gating logic 51, which varies the DSL signal according to a time sequence. The output of transmitter 50 is a gated DSL output signal, which is a time-varied representation of the original non-gated DSL transmit signal. Distortion detection and measuring logic 53 is coupled to line interface 44, as well as being correlated with the time sequence of gating logic 51. Output adjusting logic 54 is coupled to distortion detection and measuring logic 53 and to transmitter 50. Optionally (and not shown in the figures), distortion detection and measuring logic 53 may reside in central DSL device 33 of FIG. 2. However, for simplicity, the present invention is described with reference to remote DSL device 38.

Gating logic 51, distortion detection and measuring logic 53, and output adjusting logic 54 may be configured in software, hardware, firmware, or a combination thereof. For example, gating logic 51 may be implemented in hardware by such components as a modulator or a simple logic gate. Elements 51, 53 and 54 are used to implement DSP sequencing and correlation algorithms which are known in the art of data communications, and which are used to detect and measure distortion in the communications channel and to use this measurement to automatically adjust the transmit signal of DSL device 38, as set forth in the following discussion. The following discussion assumes an off-hook telephone or dial modem. However, the concepts may be applied to on-hook devices as well.

FIG. 4A illustrates an ideal frequency placement of the telephone and dial modem signals (POTS band) and the DSL modem signals (DSL band) on the two-wire communications channel 14 of FIG. 1, wherein a POTS splitter has been used. Note that there is no overlap between the POTS band signals and the DSL band signals. FIG. 4B illustrates the signal at the DSL modem 38 interface wherein the POTS band signal has been filtered by the POTS splitter. FIG. 4C illustrates the signal at the telephone 17 or dial modem 18 interface wherein the DSL signal has been filtered by the POTS splitter.

FIG. 5 illustrates the signal on the two-wire communications channel 34 of FIGS. 2 and 3, without the use of phone filters 40 and 41. In this case, the DSL signal causes non-linear distortion at the telephone 17 or dial modem 18 interface. Because the DSL signal is a time-continuous signal (that is the envelope of the DSL signal is time-stationary), the distortion products are also time-continuous and appear as wide band noise across all frequency bands. Thus, the net signal on the two-wire channel is composed of the POTS signal, the DSL signal, and the distortion signal This distortion has a direct effect on the DSL performance of DSL device 38 due to the unwanted signal components in the DSL band. The distortion more importantly is found to cause unacceptable audio noise in telephone 17 and reduces the performance of dial modem 18 due to the unwanted signal components in the dial modem signal. The distortion actually heard in a telephone speaker may exceed that indicated in FIG. 5 due to further distortion within the telephone circuitry which is blocked from appearing at the telephone interface. The distortion may also have frequency content that is different than that indicated in FIG. 5.

FIG. 6 shows the signal at the telephone 17 interface in the splitter-less DSL system of FIGS. 2 and 3 wherein optional phone filters 40 and 41 are used to reduce the amplitude of the DSL signal. In particular, the phone filter reduces the level of the DSL signal at the telephone which in turn reduces the distortion, typically by a reduction factor that is larger than the reduction factor of the DSL signal itself. As shown in FIG. 6, this reduces the level of distortion in both the POTS band and the DSL band.

With further reference to FIG. 6, it should be noted that both the DSL signal envelope and the POTS signal envelope at the dial modem 18 are time-stationary. That is, both signals are present at all times. However, a telephone voice signal is typically not time stationary because the POTS signals representing speech vary greatly in amplitude when viewed over seconds or minutes and the signal may be effectively absent. Thus, the POTS band signal may at times be absent.

FIG. 7 illustrates the signal at DSL device 38 of FIGS. 2 and 3 in the presence of the effects illustrated in FIG. 6. In FIG. 7, both the POTS band signal components produced by POTS devices 17 and 18 appear directly because the low-pass effect of phone filters 40 and 41 does not attenuate the POTS band. However, the DSL band distortion component is reduced due to the lowpass filtering of the phone filters 40 and 41 and the DSL band signal level is increased because this signal is the un-attenuated DSL signal. The distortion in the POTS band is observable at the DSL device 38. Because the communication channel 34 passes the POTS band signals and the distortion components with equal attenuation, the distortion in the POTS band is also observable at DSL device 33.

As set forth below, it is possible to detect and quantify the distortion components illustrated in FIG. 4 at DSL device 33 and DSL device 38.

The POTS band signal component due to telephone audio or due to a dial modem is not practically separable from the POTS band distortion component because the distortion component, which is originally caused by the DSL signal, is smaller in amplitude than the telephone audio or dial modem component. Thus, there is no practical ability for the DSL device to correlate the distortion component with a DSL signal so that it can be extracted. A possible exception that would allow detection would be to measure the distortion component when a telephone signal is silent. This is impractical, however, because there is no guarantee of silence and because the level of distortion to be measured is very small and is similar to the amplitude of signals due to telephone background noise.

The distortions illustrated in FIGS. 5 through 7 are due to the time stationary DSL signals. However, if the signal envelope of the DSL signals is varied with time, the DSL signals become non-stationary. The non-linear distortion caused by such non-stationary signals at a POTS device, such as a telephone or dial modem, is fundamentally different from the distortion that results from time stationary DSL signals.

When the signal envelope of the DSL signals is varied according to a time sequence, the nonlinear distortion components at the telephone or dial modem interface are correlated to that sequence. In a preferred embodiment of the invention, the DSL signals are time-varied by gating the output (transmit) signals of the DSL device on and off. Gating of the DSL signals according to a time sequence causes nonlinear distortion components at the telephone or dial modem interface that are correlated to the time sequence and in fact occur coincident in time with the gating.

Such gating can be applied to a standard DSL signal in a (short duration) distortion test mode wherein measurement takes place at the sacrifice of data communication. Alternatively, such gating may be inherent in certain DSL signaling, such as half-duplex signaling, in which case non-interruptive and adaptive measurement can take place.

As an example, consider gating that occurs at a 500 Hz rate: the DSL signal is on for some duration less than 2 msec and off for the remainder of the 2 msec, such sequence repeating continuously, as illustrated in FIG. 8A. Each burst of DSL signal, as may have been filtered by a phone filter, causes via non-linear distortion in the telephone or dial modem a momentary change in the distortion component ("DC") signal at the telephone or dial modem interface during each burst, as illustrated in FIG. 8B. (FIG. 8B represents a simplified illustration of the distortion components. Typically, the distortion components will be more complex than the DC shown in FIG. 8B. The DC value may be expected to tend toward zero due to the highpass nature of the telephone or dial modem interface or contain higher frequency components. In addition, the polarity may be reversed.)

The resultant distortion in this example differs very importantly from the distortion in the ordinary non-gated case, as illustrated with reference to FIGS. 9 and 10. As illustrated in FIG. 9, the distortion components for the 500 Hz gating example include discrete line spectra at multiples of 500 Hz. (FIG. 10 is a further simplification of FIG. 9, in which the POTS band designation has been eliminated).

Thus, wherein without gating the distortion is spread across the POTS band, with gating the distortion is concentrated at multiples of the gating frequency, which in this example is 500 Hz. The amplitude of the line spectra directly corresponds with the overall level of distortion.

It should be clear to those skilled in the art that the gating sequence may be correlated with the "DC" distortion components using conventional techniques to detect the presence and the magnitude of the distortion. For example, the magnitude of the POTS band signal at the DSL modem interface can be sampled and measured for a short time period after the beginning of every DSL signal burst, and measurement can be made over several or many bursts to accomplish correlation. This may be accomplished by using conventional components known in the art, such as a sample and hold circuit. It should also be clear that it is not necessary for the gating to be periodic. That is, even random DSL signal bursts allow detection and measurement of the distortion.

It should also be clear to those skilled in the art that such correlation techniques may be used to detect and to measure even very small levels of distortion and to do this in the presence of large POTS band signals.

As one example, immediately after an off-hook event, audio near-silence is most likely for a short time, so the background audio level should be perhaps −50 dBm, maximum. It should be possible to detect and measure distortion components above −70 dBm with a detection assurance on the order of around 12 dB. This implies correlation signal-to-noise ratio (SNR) improvement of 32 dB (40 times) is needed, requiring correlation of about 1626 samples. For 1 msec DSL bursts at a rate of 500 Hz, it is reasonable to obtain 8 independent samples per burst (0.001*8000 Hz) and thus 4000 correlation samples in one second. The above −70 dBm distortion could be measured in about 0.4 seconds.

As another example, the large distortion expected without a POTS filter with a large DSL signal level may result in distortion components at a level of at least −50 dBm, requiring correlation SNR improvement of 12 dB (16 times). This can be accomplished in perhaps 4 msec.

The above example implies that the DSL signal is both transmitted and detected at the user's DSL device, such is DSL device 38 in FIG. 2. However, the distortion is also visible at other DSL devices in the system, as is detection of the gating sequence. Thus, detection of the distortion can be accomplished at a DSL device remote from the transmitting DSL device, such as DSL device 33 in FIG. 2.

Thus, it is shown above that distortion can be detected and measured at either a local or remote DSL device. Farther, the magnitude of the distortion components in the POTS band for any given POTS-device, such as a telephone or dial modem, is directly related to the magnitude of the DSL transmit signal. This corresponding relationship is illustrated in FIGS. 11A–11C, which are simplified illustrations of three DSL device transmit signal levels and the corresponding POTS band distortion components.

For telephones, the magnitude of non-linear distortion measured in the system can be empirically matched to the levels of resulting audible noise or buzz in a given universe of telephones, and based on these findings a maximum level of tolerable distortion for the particular universe can be determined. (This determination can be made regardless of whether the universe utilizes phone filters on all telephones, does not utilize phone filters on any telephones, or uses a combination of filtered and unfiltered telephones). For purposes of this description this maximum distortion level is referred to as the target distortion or "TD". For dial modems, the TD similarly can be established to provide allowable data rate degradation of the dial modem. Because it is possible to determine if the off-hook device is a telephone or a dial modem, for example by detecting calling or answering tones, two TDs may be determined: one TD for telephones and the other TD for dial modems. Once the TD is measured, the present invention automatically and adaptively adjusts parameters of the DSL device's transmit signal to achieve the highest possible data rate consistent with the TD.

The flow chart of FIG. 12 shows the architecture, functionality, and operation of a possible implementation of the automatic adjusting software of the present invention. In this regard, each block represents a module, segment, or portion of code, which comprises one or more executable instructions for implementing the specified logical function (s). It should also be noted that in some alternative implementations, the functions noted in the blocks may occur out of the order noted in FIG. 12.

With reference now to the flow chart of FIG. 12, in a preferred embodiment the present invention functions as follows: in step 101, the transmit parameters of the DSL device are set at a level commensurate with an initial TD. In a preferred embodiment of the invention, the initial DSL device transmit parameters are set at a level commensurate with the universe of POTS-type devices in the system to provide the best initial estimate of the level adjustment required to achieve the TD. In most applications this value will correspond to the worst distortion situation expected for the universe of potential attached telephones and modems with or without phone filters (i.e., the transmit parameters must be set so that the TD is below the maximum allowable distortion fir the most distortion-sensitive POTS device in the system).

In step 103, a time-varied DSL signal is transmitted, as described above. In a preferred embodiment, the DSL signal is gated on and off according to a fixed time sequence, although the invention may be used with non-periodic and even random time sequences. In step 105, the initial POTS band distortion is measured, either at the transmitting DSL device or at a remote DSL device in the system. As discussed above, this measurement may require as little as 5–10 milliseconds.

In step 107, the transmit parameters of the DSL device are adjusted so that the measured distortion corresponds with the initial estimate of the TD. In a preferred embodiment, the adjustment is made to the power level of the DSL transmit signal. However, the TD can be achieved by adjusting other parameters of the DSL transmit signal.

In some cases, it may be desirable or necessary to repeat this process several times to achieve the desired TD, or to verify that the TD has been achieved, as illustrated in step 109. Once the DSL transmit signal parameters have been set, the signal is transmitted in step 111. The transmitted DSL signal has the highest data rate commensurate with the TD. During transmission, the distortion can be adaptively measured to determine if the actual distortion differs from the TD, as illustrated in step 113. If the distortion is found to be significantly different than the TD, the parameters of the DSL transmit signal can again be adjusted to achieve the TD, as illustrated in step 115.

It should be emphasized that the above-described embodiments of the present invention, particularly any "preferred" embodiments, are merely possible examples of implementations, merely set forth for a clear understanding of the principles of the invention. Many variations and modifications may be made to the above-described embodiment(s) of the invention without departing substantially from the spirit and principles of the invention. For example, it is possible to implement the present invention by time-varying the DSL signal in a periodic or in a random manner. Furthermore, it is possible to detect and measure the distortion either at the transmitting DSL device or at another DSL device in the system, which may be remotely located from the transmitting DSL device. All such modifications and variations are intended to be included herein within the scope of the present invention.

Therefore, having thus described the invention, at least the following is claimed:

1. A method for maximizing a DSL data rate commensurate with a known tolerable level of distortion on a communication channel to which is connected at least one conventional plain old telephone service (POTS) device and at least one digital subscriber line (DSL) device, comprising the steps of:

varying an output signal of a first DSL device by gating said output signal on and off according to a time sequence to produce a time-varied DSL signal on the communication channel;

measuring a non-linear distortion on the communication channel caused by the time-varied DSL signal a first time;

adjusting the transmit parameters of said output signal a first time based on said measured distortion; and

transmitting said adjusted output signal.

2. The method of claim 1, further comprising the steps of:

measuring the non-linear distortion a second time during said transmitting step; and

adjusting the transmit parameters of said output signal a second time based on said second measurement.

3. The method of claim 1, further comprising the steps of:

measuring the non-linear distortion a plurality of times during said transmitting step; and

adjusting the transmit parameters of said output signal a plurality of times based on said plurality of measurements.

4. The method of claim 1, wherein said gating is inherent in the output signal.

5. The method of claim 1, wherein said gating takes place during a distortion test mode.

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6. The method of claim 1, wherein said step of measuring further comprises sampling and measuring the output signal.

7. The method of claim 1, wherein said step of measuring further comprises measuring the distortion at the output of the first DSL device.

8. The method of claim 1, wherein said step of measuring further comprises measuring the distortion at a second DSL device connected to the communication channel, said second DSL device being remote from said first DSL device.

9. An apparatus for maximizing the DSL data rate commensurate with a known tolerable level of distortion on a communication channel to which is connected at least one conventional plain old telephone service (POTS) device and at least one digital subscriber line (DSL) device, comprising:

means for varying an output signal of a first DSL device by gating said output signal on and off according to a time sequence to produce a time-varied DSL signal on the communication channel;

means for measuring a non-linear distortion on the communication channel caused by the time-varied DSL signal a first time;

means for adjusting the transmit parameters of said output signal a first time based on said measured distortion; and

means for transmitting said output signal.

10. The apparatus of claim 9, wherein said varying means is a logic gate which turns the output signal on and off according to said time sequence.

11. The apparatus of claim 9, wherein said varying means varies the output signal of the first DSL device according to a random time sequence.

12. The apparatus of claim 9, wherein said varying means varies the output signal of the first DSL device according to a periodic time sequence.

13. The apparatus of claim 9, wherein said measuring means is a sample and hold circuit.

14. The apparatus of claim 9, wherein said measuring means is capable of measuring the non-linear distortion a plurality of times, and wherein said adjusting means is capable of adjusting the transmit parameters of said output signal a plurality of times based on said plurality of measurements.

15. A computer readable medium having a program for maximizing the DSL data rate commensurate with a known tolerable level of distortion on a communication channel to which is connected at least one conventional plain old telephone service (POTS) device and at least one digital subscriber line (DSL) device, the program comprising:

means for supplying an output signal of a first DSL device to a modulator, said modulator being designed to supply a modulated output signal that is varied by gating said output signal on and off according to a time sequence;

means for measuring distortion on the communication channel in the audio frequency band, said distortion being correlated with the time sequence of said modulated output signal;

means for adjusting the transmit parameters of said output signal a first time based on said measured distortion; and

means for transmitting said adjusted output signal.

16. The program of claim 15, wherein the time sequence of said modulated output signal is varied randomly.

17. The program of claim 15, wherein the time sequence of said modulated output signal is varied periodically.

18. The program of claim 15, wherein said measuring means is located at the first DSL device.

19. The program of claim 15, wherein said measuring means is located at a second DSL device connected to the communication channel, said second DSL device being remote from said first DSL device.

* * * * *

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UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 6,154,524

DATED : November 28, 2000

INVENTOR(S) : Bremer

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

On the title page,
left column, change application filing date from "January 29, 1999" to --January 28, 1999--.

Column 1, line 56, insert "signals" between --POTS-- and --so--.

Column 2, line 9, change "(transmit" to --(transmit)--.

Column 3, line 34, change "illustrating at prior" to --illustrating a prior--.

Column 3, line 64, change "that, signal" to --that signal--.

Column 8, line 64, change "such is" to --such as--.

Column 9, line 4, change "Farther" to --Further--.

Column 9, lines 37-38, change "function (s)" to --function(s)--.

Column 9, line 55, change "fir" to --for--.

Signed and Sealed this

Twenty-ninth Day of May, 2001

Attest:

Nicholas P. Godici

NICHOLAS P. GODICI

Attesting Officer

Acting Director of the United States Patent and Trademark Office

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US006111936A

United States Patent [19]
Bremer

[11] Patent Number: 6,111,936
[45] Date of Patent: Aug. 29, 2000

[54] **METHOD AND APPARATUS FOR AUTOMATICALLY DETECTING AND MEASURING DISTORTION IN A DSL SYSTEM**

[75] Inventor: **Gordon Bremer**, Clearwater, Fla.

[73] Assignee: **Paradyne Corporation**, Largo, Fla.

[21] Appl. No.: **09/239,636**

[22] Filed: **Jan. 28, 1999**

Related U.S. Application Data

[60] Provisional application No. 60/072,792, Jan. 28, 1998.

[51] **Int. Cl.**[7] **H04M 1/24**

[52] **U.S. Cl.** **379/28**; 379/6; 379/1; 379/27; 379/29

[58] **Field of Search** 379/6, 10, 12, 379/21, 22, 23, 24, 26, 27, 28, 29, 32, 1, 30, 5, 31

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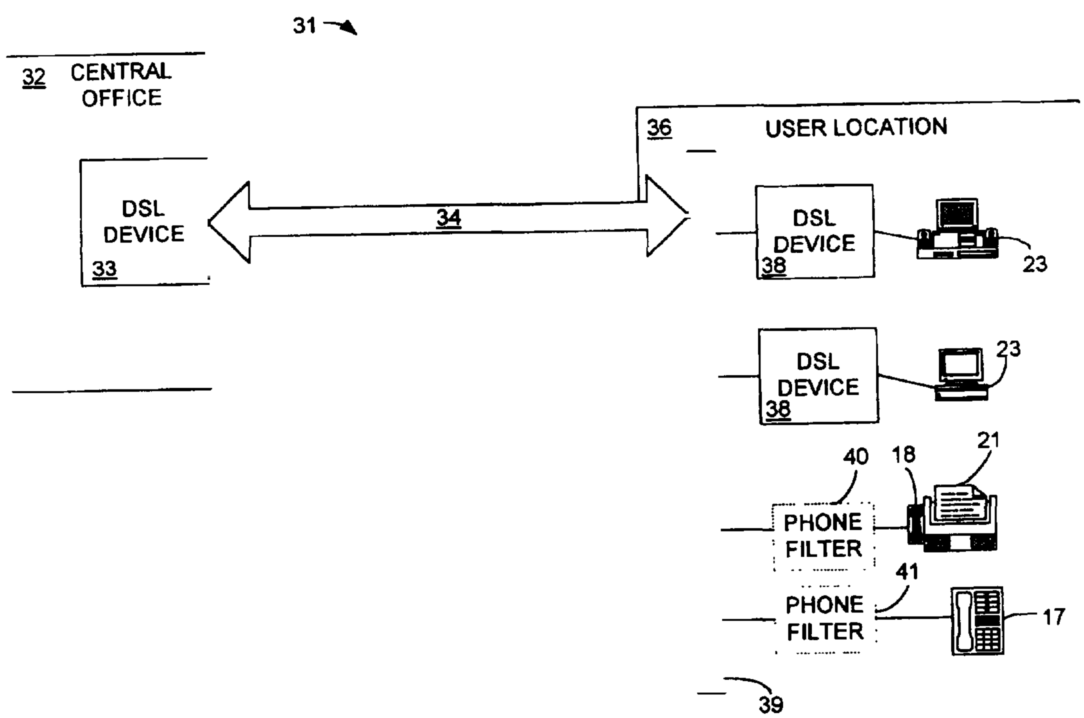
Primary Examiner—Curtis A. Kuntz
Assistant Examiner—Rexford M Barnie
Attorney, Agent, or Firm—Thomas, Kayden, Horstemeyer & Risley, L.L.P.

[57] **ABSTRACT**

In a communications environment where it is desirable to allow the simultaneous transmission of digital subscriber line (DSL) signals and conventional Plain Old Telephone Service (POTS) signals on a single two-wire communication line, a method and apparatus is configured to automatically measure the distortion caused by a telephone or other attached device such as a dial modem due to the presence of a DSL signal on the same communication line. The method and apparatus of the present invention allows DSL devices and POTS devices to be used on the same communication line without the use of a splitter.

19 Claims, 8 Drawing Sheets



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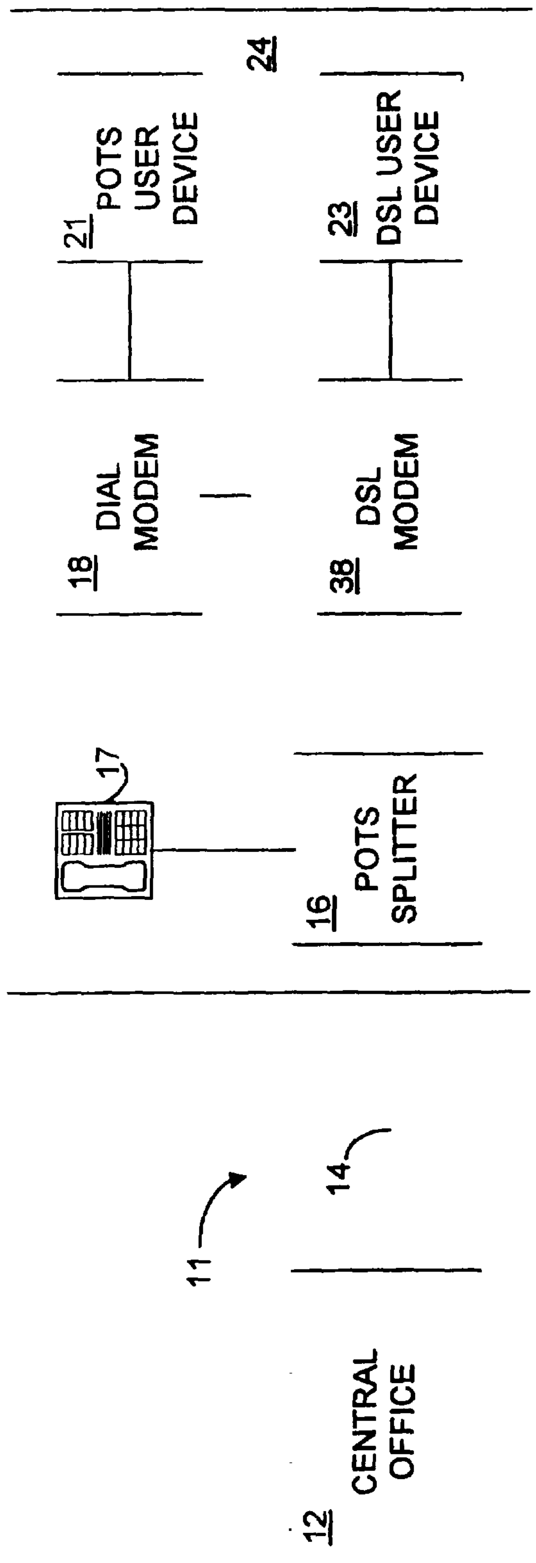

Fig. 1 (PRIOR ART)

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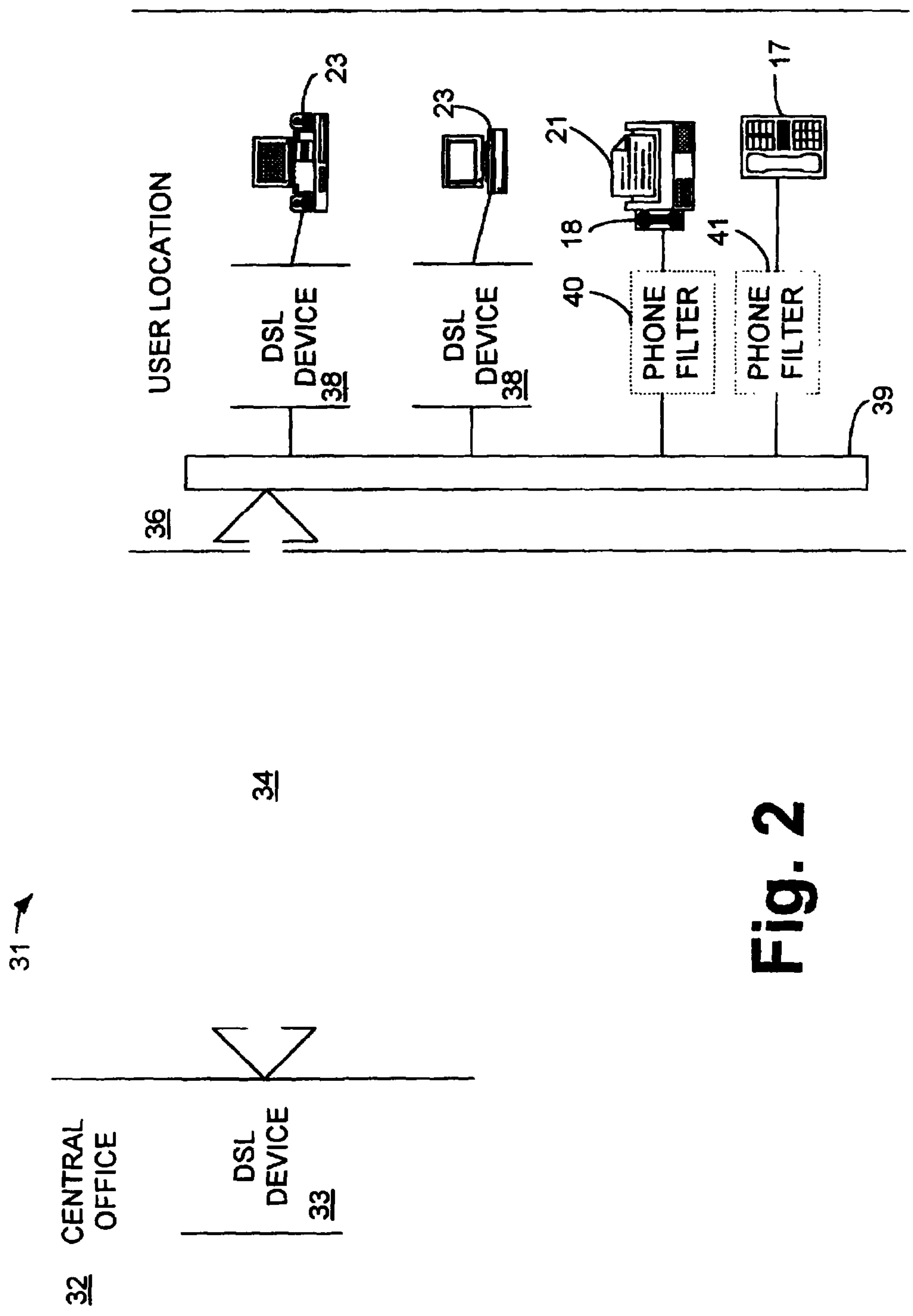
USER LOCATION
DSL DEVICE 38
DSL DEVICE 38
PHONE FILTER
PHONE FILTER
23
23
21
18
17
40
41
39
36
34
31
CENTRAL OFFICE
DSL DEVICE 33
32

Fig. 2

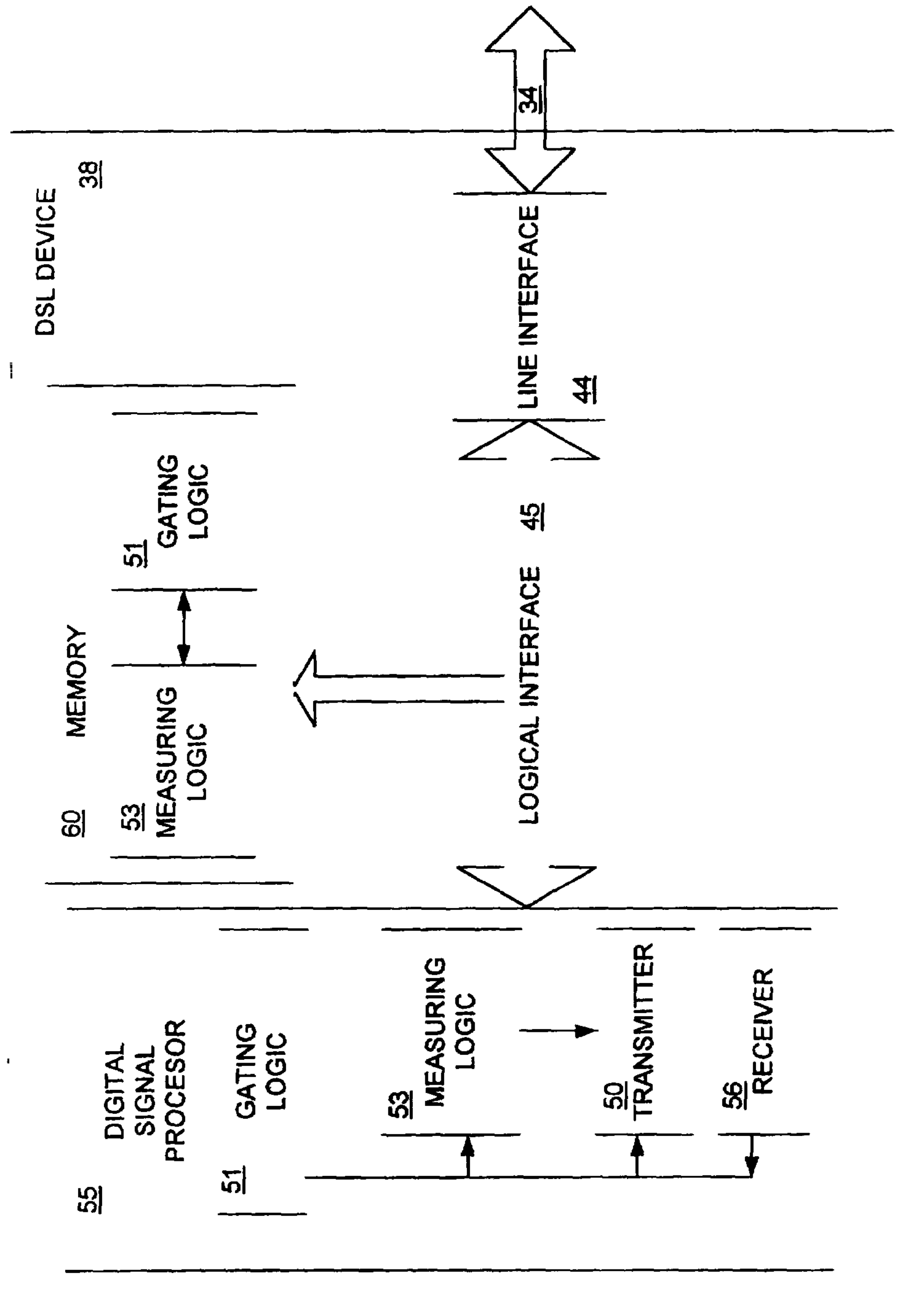
DSL DEVICE
38
34
LINE INTERFACE
44
45
LOGICAL INTERFACE
MEMORY
60
51
GATING LOGIC
53
MEASURING LOGIC
55
DIGITAL SIGNAL PROCESOR
51
GATING LOGIC
53
MEASURING LOGIC
50
TRANSMITTER
56
RECEIVER

Fig. 3

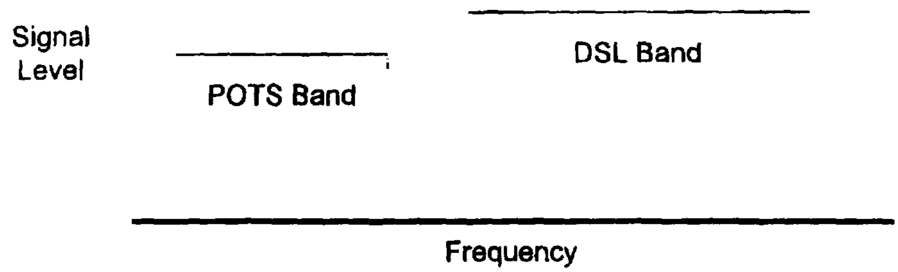

FIG. 4A

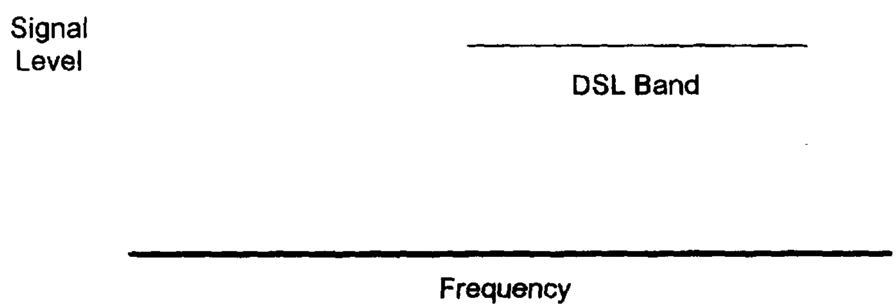

FIG. 4B

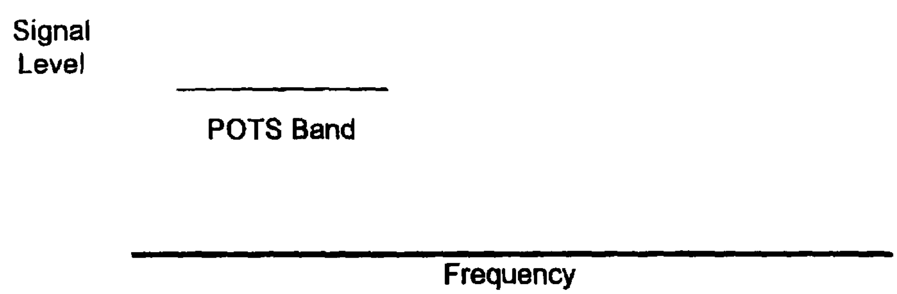

FIG. 4C

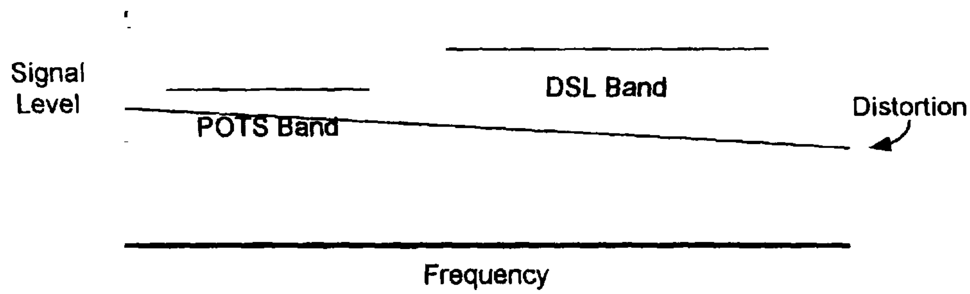

FIG. 5

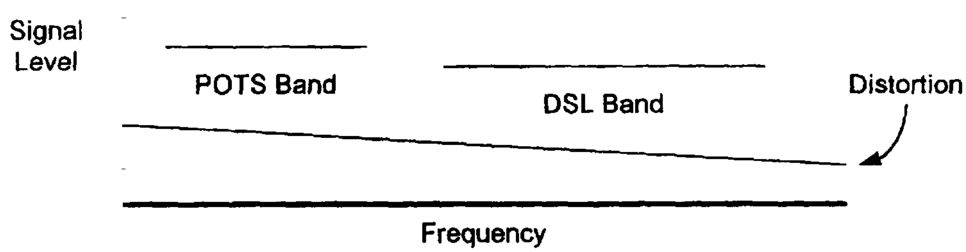

FIG. 6

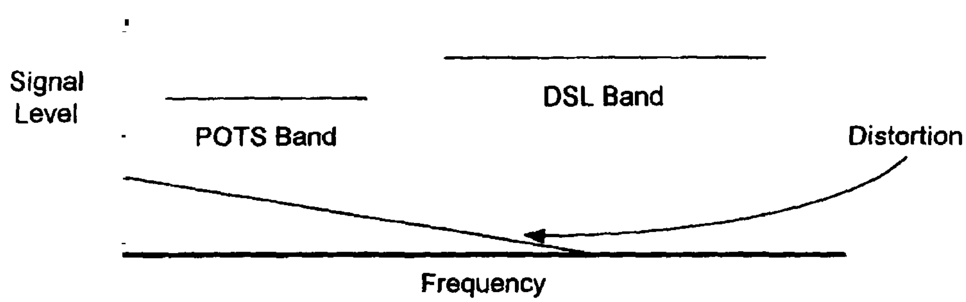

FIG. 7

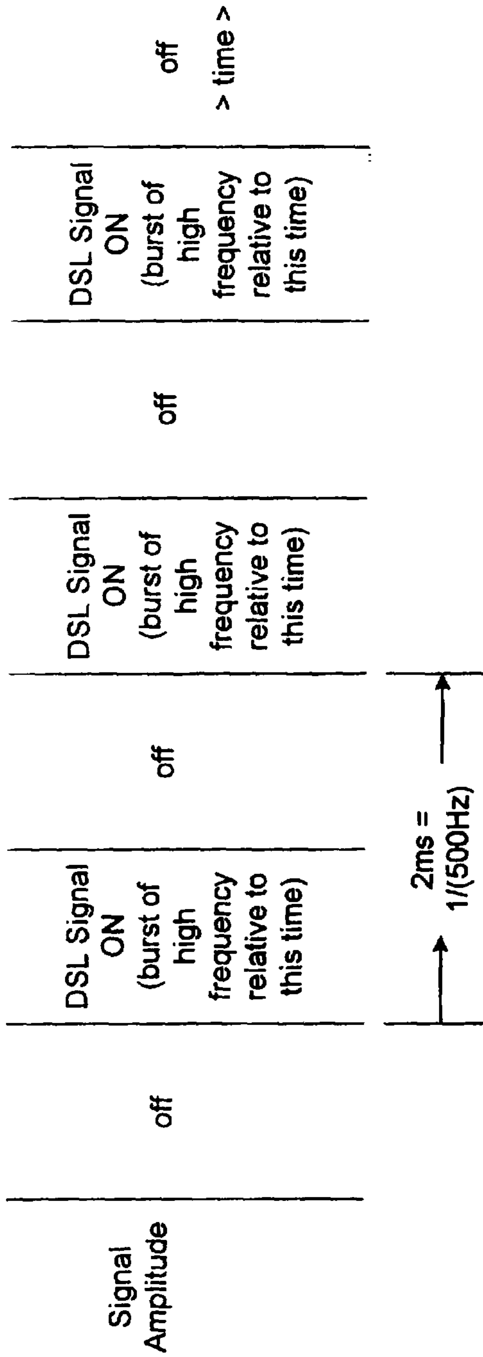
Signal Amplitude
off
DSL Signal ON (burst of high frequency relative to this time)
2ms = 1/(500Hz)
off
DSL Signal ON (burst of high frequency relative to this time)
off
DSL Signal ON (burst of high frequency relative to this time)
off
> time >

FIG. 8A

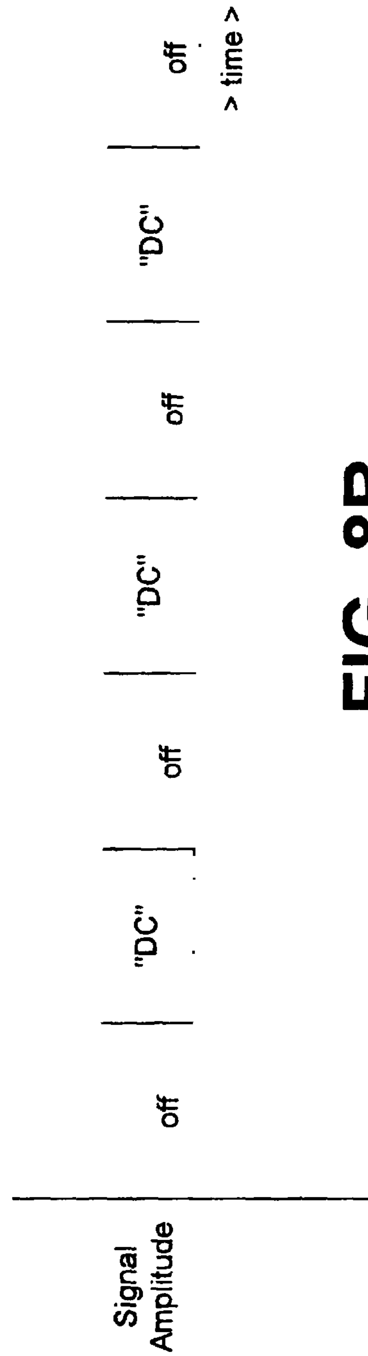
Signal Amplitude
off
"DC"
off
"DC"
off
"DC"
off
> time >

FIG. 8B

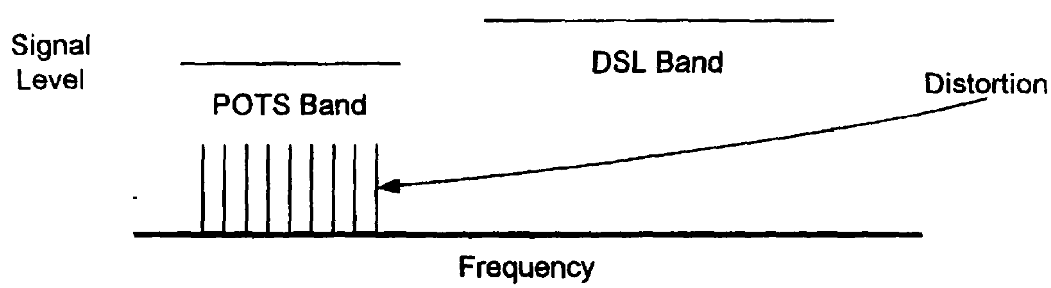

FIG. 9

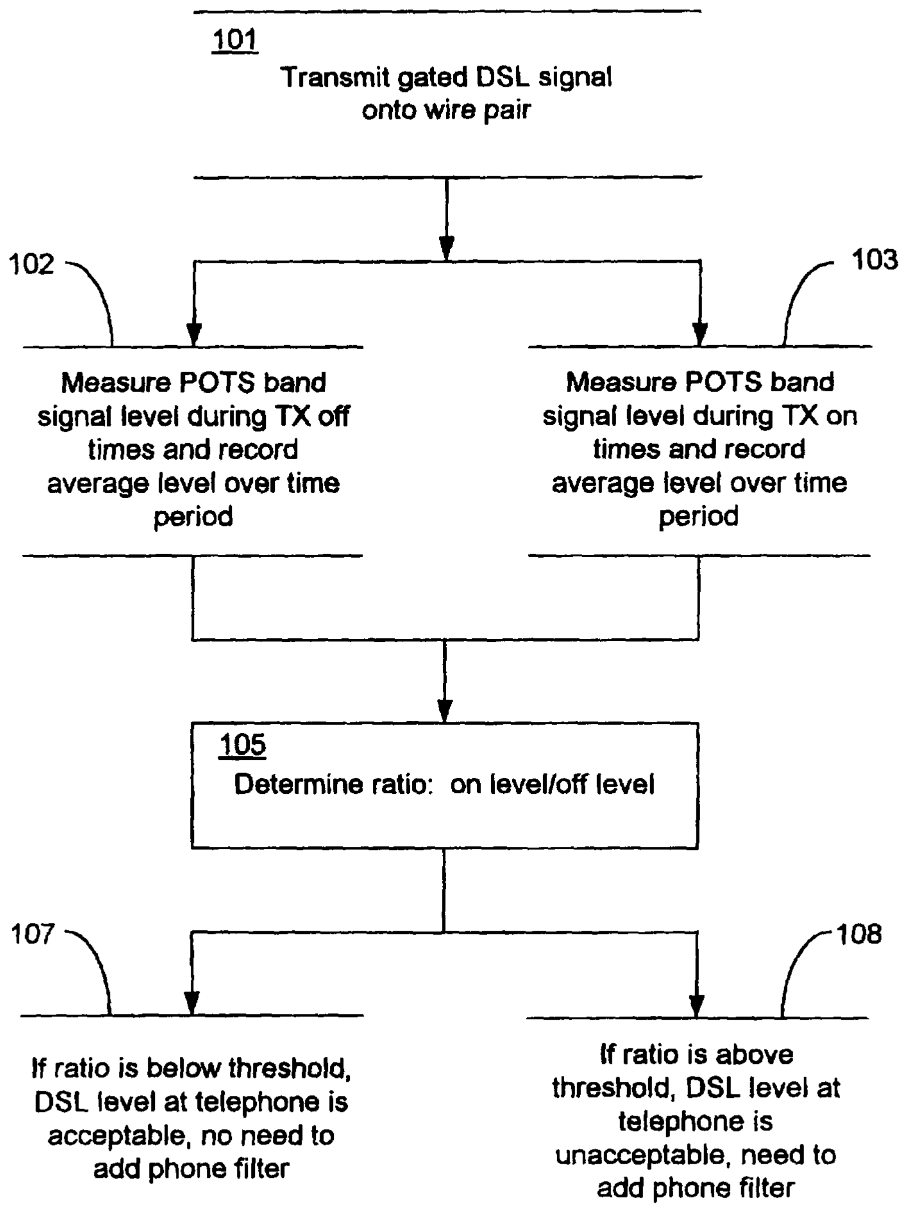
101
Transmit gated DSL signal onto wire pair
102
103
Measure POTS band signal level during TX off times and record average level over time period
Measure POTS band signal level during TX on times and record average level over time period
105
Determine ratio: on level/off level
107
108
If ratio is below threshold, DSL level at telephone is acceptable, no need to add phone filter
If ratio is above threshold, DSL level at telephone is unacceptable, need to add phone filter

FIG. 10

METHOD AND APPARATUS FOR AUTOMATICALLY DETECTING AND MEASURING DISTORTION IN A DSL SYSTEM

CROSS REFERENCE TO RELATED APPLICATIONS

This application claims priority to and the benefit of the filing date of co-pending and commonly assigned provisional application entitled AUTOMATICALLY IDENTIFYING THE NEED FOR A PHONE FILTER IN A DSL SYSTEM, assigned Ser. No. 60/072,792, filed Jan. 28, 1998, and hereby incorporated by reference. The present application is also related to copending application entitled METHOD AND APPARATUS FOR AUTOMATICALLY AND ADAPTIVELY ADJUSTING TELEPHONE AUDIO QUALITY AND DSL DATA RATE IN A DSL SYSTEM, filed on even date herewith, under express mail no. EL068409204US.

TECHNICAL FIELD

The present invention relates generally to communication devices, and more particularly, to a method and apparatus for automatically detecting and measuring distortion in a communications systems that includes digital subscriber line (DSL) signals and conventional telecommunications signals.

BACKGROUND OF THE INVENTION

In the field of data communications, modems are used to convey information from one location to another. digital subscriber line (DSL) technology now enables DSL devices, such as DSL modems, to communicate large amounts of digital data. Typically in a communications environment, Plain Old Telephone Service (POTS) type devices (such as telephones, facsimile machines, and dial modems) conveying conventional telecommunications signals are connected to the same subscriber wire pair as DSL devices at the user's location, which is typically remote from the telephone company's central office location, via a wire pair provided by the local telephone company.

Because passband DSL signals, such as asymmetric digital subscriber line (ADSL) and rate adaptive digital subscriber line (RADSL) modem signals, typically occupy only the frequency band above the audio band, these DSL signals have traditionally been isolated from all POTS type devices (such as telephones or dial modems) by a splitter or filter system installed at the user (remote) location. Such a splitter is typically known in the field of telephony communications as a POTS splitter. The POTS splitter typically serves two purposes: (1) it attenuates the DSL signals so that they do not significantly appear at the input of the POTS devices, and (2) it attenuates the POTS signals so that they do not significantly appear at the input of the DSL devices. In particular, the POTS filter attempts to attenuate DSL signals appearing at the input of the POTS devices in the audio band to an inaudible level, and also attempts to attenuate DSL signals above the audio band to a level low enough so that distortion inside the POTS type devices does not adversely affect their performance.

Although POTS splitters have been effective, such splitters are undesirable for many applications because of installation and cost issues. Recognizing the undesirable attributes of POTS splitters, efforts to eliminate them have begun in the DSL industry and standards bodies. These attempts call

for "splitter-less-DSL" systems wherein the DSL devices and the POTS devices are all directly connected to the two-wire communications channel without the use of a POTS splitter.

In splitter-less operation, it is necessary for the DSL device to: (1) filter its output signal to ensure that signals in the audio band are below an audible level, and (2) reduce its output (transmit) signal power to a level that does not cause adverse distortion in the POTS devices. The reduction in output signal power is a crucial aspect of splitter-less operation. Unfortunately, such reduction may dramatically reduce data performance, in some cases to an unacceptable level. Due to line losses, reduction in the output signal power level also reduces the reach of the subscriber loop.

To facilitate production of a splitter-less DSL device, such as a DSL modem, that can be installed in a variety of systems without field adjustment, the DSL device transmit level must be pre-set to the lowest level commensurate with the worst distortion situation expected for the potential universe of attached POTS devices. As noted above, such a reduced output level may be undesirable in many installations. Although many installations might tolerate a higher output level, this level could only be determined by trial and error for each installation, which is a cumbersome process and impractical for mass deployment of DSL splitter-less devices for consumer applications.

Due to the problems involved in using many DSL devices in splitter-less operation, it has been proposed that phone filters (also known as distributed POTS splitters) be placed at the interface to each item of POTS type equipment at the remote location. A phone filter is a bi-directional lowpass filter that attenuates as much of the DSL modem signal as practical and prevents it from appearing at the interface to the POTS type devices, such as telephones and dial modems. A phone filter can markedly improve splitter-less operation by permitting the transmit power level of the DSL device to be increased above the level that would produce distortion in splitter-less operation. However, phone filters also have cost and installation drawbacks and they are less suitable for some types of DSL operation than a system utilizing a POTS splitter. Thus, it is desirable to eliminate phone filters where they are not required, for example on a particular POTS device that is immune to distortion.

Therefore, there is a need in the industry for a method and apparatus for determining at a DSL device whether the distorting effects of the DSL device's transmit signal warrant the incorporation of a phone filter at the interface to each POTS device in the system. There is also a need to be able to make such determinations at both the user (remote) end and at the central site end of the two-wire communications channel.

SUMMARY OF THE INVENTION

A technique is presented that permits a DSL-type device on a wire-pair communication channel to automatically identify a POTS-type device (such as a telephone or dial modem) on that same wire pair that, in the presence of a DSL-type line signal, is causing distortion either: (1) detrimental to the data performance of the DSL-type device; (2) likely to be detrimental to the audio quality of the telephone and telephone system including the introduction of audio noise; or (3) detrimental to the data rate performance of the dial modem. Such identification may be used to denote the need for placement of a phone filter in the wire pair prior to the POTS-type devices, which would reduce the detrimental effects of the DSL signal.

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The technique utilizes the fact that certain envelope-modulated or time-gated DSL-type passband signals cause certain predicable audio frequency band signals on the wire pair connected to a distorting POTS-type device, such as a telephone or dial modem, and that these audio frequency band signals can be detected and quantified in the DSL-type device, even in the presence of other audio signals that ordinarily occur on the wire pair channel. The technique is implemented by varying the output signal of a DSL-type device on the wire pair channel according to a time sequence to produce a time-varied DSL signal, measuring the audio frequency band distortion on the communication channel in the presence of the time-varied DSL signal, and correlating the audio frequency band distortion on the wire pair channel to the time sequence of the time-varied DSL signal.

BRIEF DESCRIPTION OF THE SEVERAL VIEWS OF THE DRAWINGS

The present invention, as defined in the claims, can be better understood with reference to the following drawings. The drawings are not necessarily to scale, emphasis instead being placed on clearly illustrating the principles of the present invention.

FIG. 1 is a schematic view illustrating a prior art communications environment in which a POTS splitter is employed to isolate a DSL device from conventional POTS-type devices;

FIG. 2 is a schematic view of a multipoint communications environment including DSL devices employing the automatic distortion detection and measurement method and apparatus of the present invention;

FIG. 3 is a simplified schematic view of a DSL device of FIG. 2 employing the automatic distortion detection method and apparatus of the present invention;

FIGS. 4A–4C are simplified illustrations of the signal level and frequency placement of POTS band signals and DSL band signals on the two-wire communications channel of FIG. 1;

FIG. 5 is a simplified illustration of the signal level and frequency placement of POTS band signals, DSL band signals, and distortion on the two-wire communications channel of FIG. 2, in which a phone filter is not used;

FIG. 6 is a simplified illustration of the signals of FIG. 5 measured at the conventional telephone interface, where a phone filter has been used to reduce the level of the DSL signal, and hence the distortion;

FIG. 7 is a simplified illustration of the signals of FIG. 5 measured at the DSL device, where a phone filter has been used to reduce the level of the DSL signal, and hence the distortion;

FIGS. 8A and 8B are simplified illustrations of the correlation of the DSL signal and the distortion components of that signal when the DSL signal is time-varied according to the present invention;

FIG. 9 is a simplified illustration of the signal level and frequency placement of POTS band signals, DSL band signals, and distortion on the two-wire communications channel of FIG. 2 when the DSL signal is time-varied according to the present invention; and

FIG. 10 is a flow chart illustrating the operation of the method and apparatus of the present invention for automatically detecting and measuring distortion in a DSL system in order to determine whether the distortion is within acceptable levels.

DETAILED DESCRIPTION OF THE INVENTION

The present invention can be implemented in software, hardware, firmware, or a combination thereof. In the preferred embodiment, the elements of the present invention are implemented in software that is stored in a memory and that configures and drives a suitable digital signal processor (DSP) situated in the respective DSL-type device. However, the foregoing software can be stored on any computer-readable medium for use by or in connection with any suitable computer-related system or method. In the context of this document, a computer-readable medium is 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 (CD-ROM) (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.

FIG. 1 is a schematic view illustrating a prior art communications environment 11. Communications channel 14 is a conventional two-wire communications system that typically connects a telephone company central office location 12 to a remote user location 24. Remote user location 24 is typically a residential or business location and includes POTS devices such as telephone 17, dial modem 18, and user device 21, as well as DSL devices, such as DSL modem 38 and user device 23. A POTS splitter 16 is employed to connect the telephone 17, dial modem 18, and DSL modem 38 to the communications channel 14. POTS splitter 16 is required in this application in order to isolate the telephone 17 and dial modem 18 from the DSL modem 38. POTS splitter 16 is typically located at remote location 24.

The POTS splitter 16 typically serves two purposes: (1) it attenuates the DSL signals so that they do not significantly appear at the input of the telephone 17 or dial modem 18, and (2) it attenuates the telephone 17 or dial modem 18 transmit signals so that they do not significantly appear at the input of the DSL modem 38. In particular, the POTS splitter 16 attempts to attenuate the DSL modem 38 signals appearing at the telephone 17 or dial modem 18 input in the audio band to an inaudible level, and also attempts to attenuate DSL modem 38 signals above the audio band to a level low enough so that distortion in the telephone 17 or dial is modem 18 does not adversely affect their performance.

FIG. 2 is a schematic view of a multipoint communications environment 31 in which POTS devices 17 and 18, and DSL devices 33 and 38 operate. In this figure, the DSL devices 33 and 38 employ the automatic distortion detection and measurement method and apparatus of the present invention. Notably, communications environment 31 does not include a POTS splitter. Optional phone filters 40 and 41 may be used to reduce distortion, as discussed hereinafter.

User location 36 is connected to central office location 32 via communication channel 34. DSL device 33 is located at central office location 32, while at least one DSL device 38 is located at the user location 36, which is remote from central office location 32. Communication channel 34 is typically the two-wire communication channel that extends between a telephone company central office and a remote residential, business, or any other location served by local telephone service. Remote location 36 may contain a plurality of remote DSL devices 38 connecting a plurality of user devices 23 to communication channel 34 via communication bus 39. Communication bus 39 is illustratively the wiring infrastructure used throughout a remote location to connect remote DSL devices 38 to communication channel 34. In addition, remote location 36 may contain conventional POTS devices 17 and 18. Illustratively, conventional telephone 17 and dial modem 18 (which is not shown in FIG. 2, but is included within user device 21, which is illustratively a facsimile machine) are connected to communication bus 39, and thus to communication channel 34. By using DSL device 33 and remote DSL devices 38 employing the concepts and features of the automatic distortion detection and measuring apparatus and method of the present invention, it is possible in some instances to connect POTS devices 17 and 18 directly to communication channel 34 without experiencing audible distortion. If the distortion detection and measurement method and apparatus of the present invention indicates an unacceptable level of distortion at the output of DSL devices 38, optional phone filters 40 and 41 may be used to reduce the distortion at POTS devices 17 and 18 to acceptable levels.

Now referring to FIG. 3, shown is a schematic view illustrating DSL device 38 of FIG. 2, including the distortion detection and measuring logic 53 of the present invention. Typically, DSL device 38 will transmit signals to the central office 32 of FIG. 2 over communication channel 34. Similarly, central office 32 will transmit signals to DSL device 38. DSL device 38 contains conventional components as are known in the art of data communications. Digital signal processor (DSP) 55 controls the operation of and includes transmitter 50 and receiver 56 of DSL device 38. DSP 55 couples through logical interface 45 to line interface 44 to gain access to communication channel 34. Also included in DSP 55 of DSL device 38 are gating logic 51, and distortion detection and measuring logic 53, which enable DSL device 38 to perform the distortion detection and measurement functions of the present invention, as discussed hereinbelow. Also contained within DSL device 38 is memory 60, which also includes gating logic 51, and distortion detection and measuring logic 53. In a preferred embodiment, the logic of the present invention is executed within DSP 55 and is therefore shown as residing in both memory 60 and DSP 55.

Still referring to FIG. 3, the output (transmit) DSL signal of DSL device 38 is passed to communication channel 34 (via logical interface 45 and line interface 44) from transmitter 50. Transmitter 50 is controlled by gating logic 51, which varies the DSL signal according to a time sequence. The output of transmitter 50 is a gated DSL output signal, which is a time-varied representation of the original non-gated DSL transmit signal. Distortion detection and measuring logic 53 is coupled to line interface 44, as well as being correlated with the time sequence of gating logic 51. Optionally (and not shown in the figures), distortion detection and is measuring logic 53 may also reside in central DSL device 33 of FIG. 2. However, for simplicity, the present invention is described with reference to remote DSL device 38.

Gating logic 51, and distortion detection and measuring logic 53, may be configured in software, hardware, firmware, or a combination thereof. For example, gating logic 51 may be implemented in hardware by such components as a modulator or a simple logic gate. Elements 51 and 53 are used to implement DSP sequencing and correlation algorithms which are known in the art of data

communications, and which are used to detect and measure distortion in the communications channel, as set forth in the following discussion. The following discussion assumes an off-hook telephone or dial modem. However, the concepts may be applied to on-hook devices as well.

The following discussion illustrates how the apparatus of FIGS. 2 and 3 is used to detect and measure distortion in the communications channel.

FIG. 4A illustrates an ideal frequency placement of the telephone and dial modem signals (POTS band) and the DSL modem signals (DSL band) on the two-wire communications channel 14 of FIG. 1, wherein a POTS splitter has been used. Note that there is no overlap between the POTS band signals and the DSL band signals. FIG. 4B illustrates the signal at the DSL modem 38 interface wherein the POTS band signal has been filtered by the POTS splitter. FIG. 4C illustrates the signal at the telephone 17 or dial modem 18 interface wherein the DSL signal has been filtered by the POTS splitter.

FIG. 5 illustrates the signal on the two-wire channel of the communications channel 34 of FIGS. 2 and 3, without the use of phone filters 40 and 41. In this case, the DSL signal causes non-linear distortion at the telephone 17 or dial modem 18 interface. Because the DSL signal is a time-continuous signal (that is the envelope of the DSL signal is time-stationary), the distortion products are also time-continuous and appear as wide band noise across all frequency bands. Thus, the net signal on the two-wire channel is composed of the POTS signal, the DSL signal, and the distortion signal. This distortion has a direct effect on the DSL performance of DSL device 38 due to the unwanted signal components in the DSL band. The distortion more importantly is found to cause unacceptable audio noise in telephone 17 and reduces the performance of dial modem 18 due to the unwanted signal components in the dial modem signal. The distortion actually heard in a telephone speaker may exceed that indicated in FIG. 5 due to even further distortion within the telephone circuitry which is blocked from appearing at the telephone interface. The distortion also may have frequency content that is different than that indicated in FIG. 5.

FIG. 6 shows the signal at the telephone 17 interface in the splitter-less DSL system of FIGS. 2 and 3 wherein optional phone filters 40 and 41 are used to reduce the amplitude of the DSL signal. In particular, the phone filter reduces the level of the DSL signal at the telephone which in turn reduces the distortion, typically by a reduction factor that is larger than the reduction factor of the DSL signal itself. As shown in FIG. 6, this reduces the level of distortion in both the POTS band and the DSL band.

With further reference to FIG. 6, it should be noted that both the DSL signal envelope and the POTS signal envelope at the dial modem 18 are time-stationary. That is, both signals are present at all times. However, a telephone voice signal is typically not time stationary because the POTS signals representing speech vary greatly in amplitude when viewed over seconds or minutes and the signal may be effectively absent. Thus, the POTS band signal may at times be absent.

FIG. 7 illustrates the signal at DSL device 38 of FIGS. 2 and 3 in the presence of the effects illustrated in FIG. 6. In FIG. 7, both the POTS band signal components produced by POTS devices 17 and 18 appear directly because the low-pass effect of phone filters 40 and 41 does not attenuate the POTS band. However, the DSL band distortion component is reduced due to the lowpass filtering of the phone filters 40 and 41 and the DSL band signal level is increased because this signal is the unattenuated DSL signal. The distortion in the POTS band is observable at the DSL device 38. Because the communication channel 34 passes the POTS band signals and the distortion components with equal attenuation, the distortion in the POTS band is also observable at DSL device 33.

As set forth below, it is possible to detect and quantify the distortion components illustrated in FIG. 4 at DSL device 33 and DSL device 38.

The POTS band signal component due to telephone audio or due to a dial modem is not practically separable from the POTS band distortion component because the distortion component, which is originally caused by the DSL signal, is smaller in amplitude than the telephone audio or dial modem component. Thus, there is no practical ability for the DSL device to correlate the distortion component with a DSL signal so that it can be extracted. A possible exception that would allow detection would be to measure the distortion component when a telephone signal is silent. This is impractical, however, because there is no guarantee of silence and because the level of distortion to be measured is very small and not dissimilar to the amplitude of signals due to telephone background noise.

The distortions illustrated in FIGS. 5 through 7 are due to the time stationary DSL signals. However, if the signal envelope of the DSL signals is varied with time, the DSL signals become non-stationary. The non-linear distortion caused by such non-stationary signals at a POTS device, such as a telephone or dial modem, is fundamentally different from the distortion that results from time stationary DSL signals.

When the signal envelope of the DSL transmit signals is varied according to a time sequence (by gating logic 51 of FIG. 3), the nonlinear distortion components at the telephone or dial modem interface are correlated to that sequence. In a preferred embodiment of the invention, the DSL transmit signals are time-varied by gating the output (transmit) signals of the DSL device on and off. Gating of the DSL signals according to a time sequence causes nonlinear distortion components at the telephone or dial modem interface that are correlated to the time sequence and in fact occur coincident in time with the gating.

Such gating can be applied to a standard DSL signal in a (short duration) distortion test mode wherein measurement takes place at the sacrifice of data communication. Alternatively, such gating may be inherent in certain DSL signaling, such as half duplex signaling, in which case non-interruptive and adaptive measurement can take place.

As an example, consider gating that occurs at a 500 Hz rate: the DSL signal is on for some duration less than 2 msec and off for the remainder of the 2 msec, such sequence repeating continuously, as illustrated in FIG. 8A. Each burst of DSL signal, as may have been filtered by a phone filter, causes via non-linear distortion in the telephone or dial modem a momentary change in the distortion component ("DC") signal at the telephone or dial modem interface during each burst, as illustrated in FIG. 8B. (FIG. 8B represents a simplified illustration of the distortion components. Typically, the distortion components will be more complex than the DC shown in FIG. 8B. The DC value may be expected to tend toward zero due to the highpass nature of the telephone or dial modem interface or contain higher frequency components. In addition, the polarity may be reversed.)

The resultant distortion in this example differs very importantly from the distortion in the ordinary non-gated case, as illustrated with reference to FIG. 9. As illustrated in FIG. 9, the distortion components for the 500 Hz gating example include discrete line spectra at multiples of 500 Hz. Thus, wherein without gating the distortion is spread across the POTS band, with gating the distortion is concentrated at multiples of the gating frequency, which in this example is 500 Hz. The amplitude of the line spectra directly corresponds with the overall level of distortion.

It should be clear to those skilled in the art that the gating sequence may be correlated with the "DC" distortion components using conventional techniques to detect the presence and the magnitude of the distortion. For example, the magnitude of the POTS band signal at the DSL modem interface can be sampled and measured for a short time period after the beginning of every DSL signal burst, and measurement can be made over several or many bursts to accomplish correlation. It should also be clear that it is not necessary for the gating to be periodic. That is, even random DSL signal bursts allow detection and measurement of the distortion.

It should also be clear to those skilled in the art that such correlation techniques may be used to detect and to measure even very small levels of distortion and to do this in the presence of large POTS band signals.

As one example, immediately after an off-hook event, audio near-silence is most likely for a short time, so the background audio level should be perhaps −50 dBm, maximum. It should be possible to detect and measure distortion components above −70 dBm with a detection assurance on the order of around 12 dB. This implies correlation signal-to-noise ratio (SNR) improvement of 32 dB (40 times) is needed, requiring correlation of about 1626 samples. For 1 msec DSL bursts at a rate of 500 Hz, it is reasonable to obtain 8 independent samples per burst (0.001 * 8000 Hz) and thus 4000 correlation samples in one second. The above −70 dBm distortion could be measured in about 0.4 seconds.

As another example, the large distortion expected without a POTS filter with a large DSL signal level may result in distortion components at a level of at least −50 dBm, requiring correlation SNR improvement of 12 dB (16 times). This can be accomplished in perhaps 4 msec.

The above example implies that the DSL signal is both transmitted and detected at the user's DSL device, such as DSL device 38 in FIG. 2. However, the distortion is also visible at other DSL devices in the system, as is detection of the gating sequence. Thus, detection of the distortion can be accomplished at a DSL device remote from the transmitting DSL device, such as DSL device 33 in FIG. 2.

The flow chart of FIG. 10 shows the architecture, functionality, and operation of a possible implementation of the automatic distortion detection and measuring software of the present invention. In this regard, each block represents a module, segment, or portion of code, which comprises one or more executable instructions for implementing the specified logical function(s). It should also be noted that in some alternative implementations, the functions noted in the blocks may occur out of the order noted in FIG. 10.

As illustrated by the flow chart of FIG. 10, in a preferred embodiment the present invention functions as follows: in step 101, the DSL device transmits a gated DSL signal onto the communication channel. In step 102, the POTS band signal level is measured at the DSL device during the time in which the DSL signal is gated "off", and the average POTS band signal level during the "off" period is recorded. Similarly, in step 103, the POTS band signal level is measured at the DSL device during the time period in which

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the DSL signal is gated "on", and the average POTS band signal level during the "on" period is recorded.

Next, in step **105** the ratio of the POTS band signal level during the "on" period to the POTS band signal level during the "off" period is determined. The ratio determined in step **105** represents the degree to which the POTS band distortion is higher during DSL transmission. In steps **107** and **108**, the ratio determined in step **105** is compared to a threshold level of acceptable audio noise, which is determined empirically, such as by audio sound measurements. This threshold indicates the ratio above which unacceptable distortion is expected from the POTS devices in the presence of the DSL transmit signal.

As illustrated in step **107**, if the ratio determined in step **105** is below the threshold, the distortion caused by the DSL level at the telephone is acceptable, and there is no need to add a phone filter. On the other hand, as illustrated in step **108**, if the ratio determined in step **105** is above the threshold, the distortion caused by the DSL level at the telephone is unacceptable, which indicates the need to add a phone filter.

It should be noted that if the DSL is not half-duplex, then the measurement illustrated in FIG. **10** will disrupt normal data communication, so it will be necessary to enter a test mode to make the measurements. If the DSL is half-duplex, communication need not be interrupted, although it may be preferable to enter a test mode in order to reduce test time. In either case, when the test mode is entered, the DSL is suitably configured and the appropriate measurements are made

Thus, it is shown above that distortion can be detected and measured at either a local or remote DSL device to determine whether the distorting effects of the DSL device transmit signal warrant the incorporation of a phone filter at the interface to each POTS-type device in the system.

It should be emphasized that the above-described embodiments of the present invention, particularly any "preferred" embodiments, are merely possible examples of implementations, merely set forth for a clear understanding of the principles of the invention. Many variations and modifications may be made to the above-described embodiment(s) of the invention without departing substantially from the spirit and principles of the invention. For example, it is possible to implement the present invention by time-varying the DSL signal in a periodic or in a random manner. Furthermore, it is possible to detect and measure the distortion either at the transmitting DSL device or at another DSL device in the system, which may be remotely located from the transmitting DSL device. All such modifications and variations are intended to be included herein within the scope of the present invention.

Therefore, having thus described the invention, at least the following is claimed:

1. A method for determining distortion on a communication channel to which is connected at least one conventional Plain Old Telephone Service (POTS) device and at least one digital subscriber line (DSL) device, comprising the steps of:

varying an output signal of a first DSL device by gating said output signal on and off according to a time sequence to produce a time-varied DSL signal on the communication channel;

measuring an audio frequency band distortion on the communication channel; and

correlating the audio frequency band distortion on the communication channel to the time sequence of the time-varied DSL signal.

2. The method of claim **1**, wherein the time sequence in said varying step is random.

3. The method of claim **1**, wherein the time sequence in said varying step is periodic.

4. The method of claim **1**, further comprising the steps of:

determining a ratio of the magnitude of the audio frequency band distortion measured during such time as the output signal is gated on to the magnitude of the audio frequency band distortion during such time as the output signal is gated off; and

comparing said ratio to a known threshold level of audio frequency band distortion.

5. The method of claim **1**, wherein said gating is inherent in the output signal.

6. The method of claim **1**, wherein said gating takes place during a distortion test mode.

7. The method of claim **1**, wherein said step of measuring further comprises sampling and measuring the output signal.

8. The method of claim **1**, wherein said step of measuring further comprises measuring the distortion at the output of the first DSL device.

9. The method of claim **1**, wherein said step of measuring further comprises measuring the distortion at a second DSL device connected to the communication channel, said second DSL device being remote from said first DSL device.

10. An apparatus for determining distortion on a communication channel to which is connected at least one POTS device and at least one DSL device, comprising:

means for varying an output signal of a first DSL device by gating said output signal on and off according to a time sequence to produce a time-varied DSL output signal on the communication channel;

means for measuring an audio frequency distortion on the communication channel;

means for coupling said measuring means to said varying means; and

means for correlating the measured audio frequency distortion with the time sequence of said time-varied DSL output signal.

11. The apparatus of claim **10**, wherein said varying means is a logic gate which turns the output signal on and off according to said time sequence.

12. The apparatus of claim **10**, wherein said varying means varies the output signal of the first DSL device according to a random time sequence.

13. The apparatus of claim **10**, wherein said varying means varies the output signal of the first DSL device according to a periodic time sequence.

14. The apparatus of claim **10**, wherein said measuring means is a sample and hold circuit.

15. A computer readable medium having a program for determining distortion on a communication channel to which is connected at least one POTS device and at least one DSL device, the program comprising:

means for varying an output signal of a first DSL device by gating said output signal on and off according to a time sequence to produce a time-varied DSL signal on the communication channel;

means for measuring distortion on the communication channel in the audio frequency band; and

means for correlating the distortion measured by said measuring means with the time sequence of said time-varied DSL signal.

16. The program of claim **15**, wherein the time sequence of said time-varied DSL signal is varied randomly.

17. The program of claim **15**, wherein the time sequence of said time-varied DSL signal is varied periodically.

18. The program of claim 15, wherein said measuring means is located at the first DSL device.

19. The program of claim 15, wherein said measuring means is located at a second DSL device connected to the communication channel, said second DSL device being remote from said first DSL device.

* * * * *

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 6,111,936
DATED : August 29, 2000
INVENTOR(S) : Bremer

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

a) Column 1, line 26, change "systems" to --system--.

b) Column 1, line 32, change "digital" to --Digital--.

c) Column 4, lines 50-51, change "dial is modem" to --dial modem--.

d) Column 5, lines 56-57, change "detection and is measuring logic" to --detection and measuring logic--.

Signed and Sealed this

Twenty-ninth Day of May, 2001

Attest:

Nicholas P. Godici

NICHOLAS P. GODICI

Attesting Officer *Acting Director of the United States Patent and Trademark Office*

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US006075784A

United States Patent [19]

Frankel et al.

[11] **Patent Number: 6,075,784**

[45] **Date of Patent: Jun. 13, 2000**

[54] **SYSTEM AND METHOD FOR COMMUNICATING VOICE AND DATA OVER A LOCAL PACKET NETWORK**

[75] Inventors: **David P. Frankel**; **Joe Boucher**, both of Sunnyvale; **Kenneth M. Kolderup**, San Mateo, all of Calif.

[73] Assignee: **Jetstream Communications, Inc.**, San Jose, Calif.

[21] Appl. No.: **09/112,911**

[22] Filed. **Jul. 9, 1998**

Related U.S. Application Data

[60] Provisional application No. 60/088,399, Jun. 9, 1998.

[51] **Int. Cl.**^{7} **H04L 12/66**

[52] **U.S. Cl.** **370/356**; 370/352

[58] **Field of Search** 370/352, 353, 370/354, 356, 355, 466, 467; 455/3.1, 3.2, 3.3, 4.1, 4.2, 5.1, 6.3; 379/88.17; 348/6, 7, 8, 12, 13, 14, 15, 16

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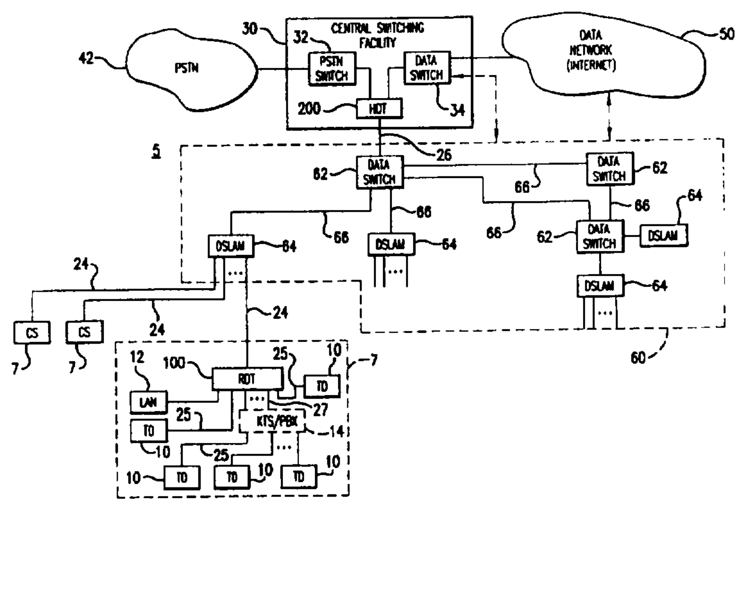
Attorney, Agent, or Firm—Needle & Rosenberg, P.C.

[57] **ABSTRACT**

A system and method of communicating voice and data via a local packet network (LPN) to and from a customer site A remote digital terminal (RDT) is provided at a customer site to interface a plurality of telephone devices and/or data devices (computers or a local area network of computers) with the LPN via a local loop link, such as a Digital Subscriber Line or a wireless local loop. A host digital terminal (HDT) is provided at a control site within or connected to the LPN that coordinates the communication of voice calls between the RDT and a public switched telephone network (PSTN) switch via the LPN and that coordinates the communication of data between the RDT and a data network within or without the LPN. Multiple telephone calls with the customer site can be supported by the remote digital terminal over a single local loop link connected to the LPN. In addition, the HDT can support communication with multiple RDTs.

26 Claims, 6 Drawing Sheets



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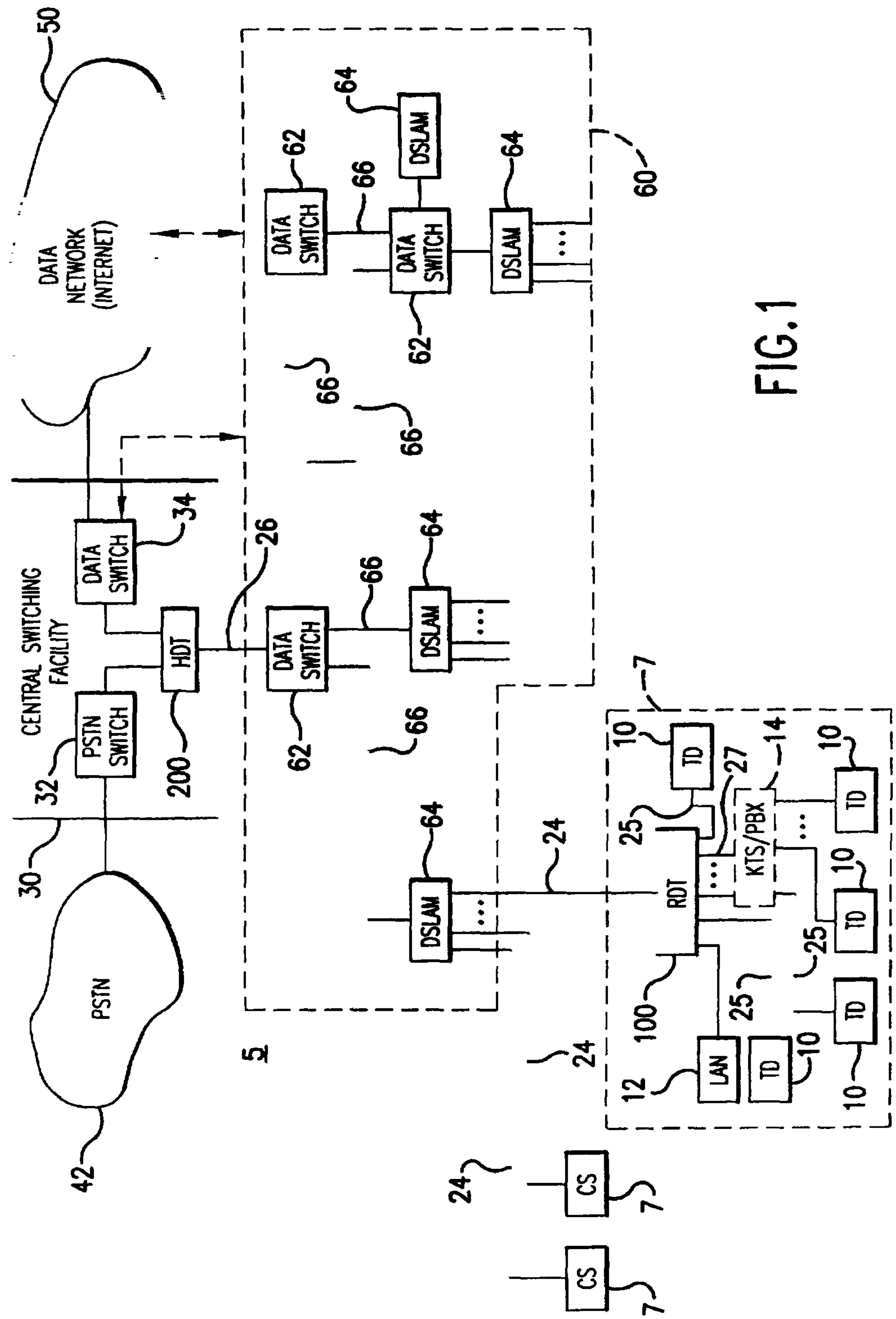

FIG. 1

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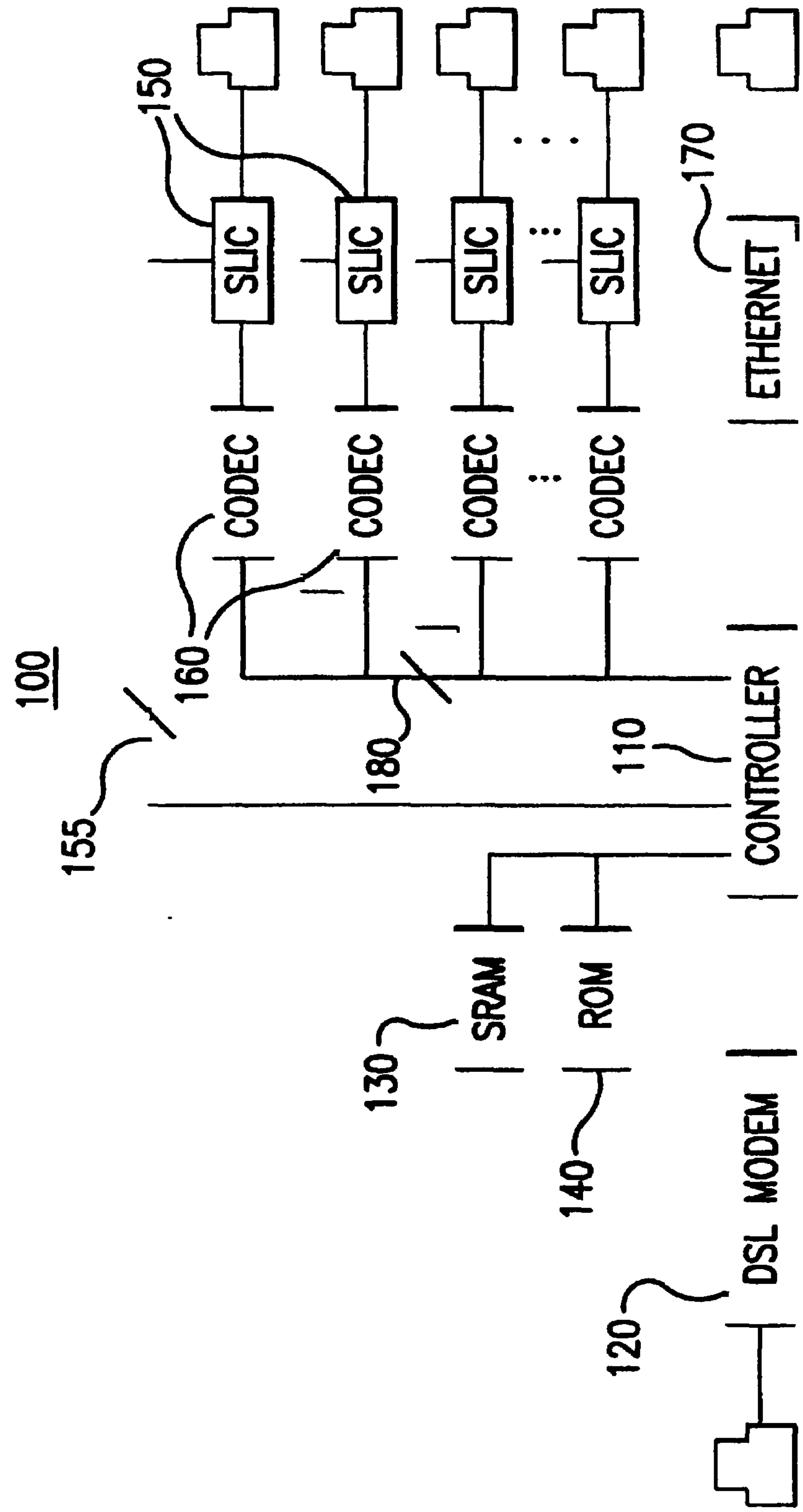
100
150
SLIC
SLIC
SLIC
SLIC
170
ETHERNET
CODEC
CODEC
CODEC
CODEC
160
155
180
110
CONTROLLER
SRAM
ROM
130
140
DSL MODEM
120

FIG.2

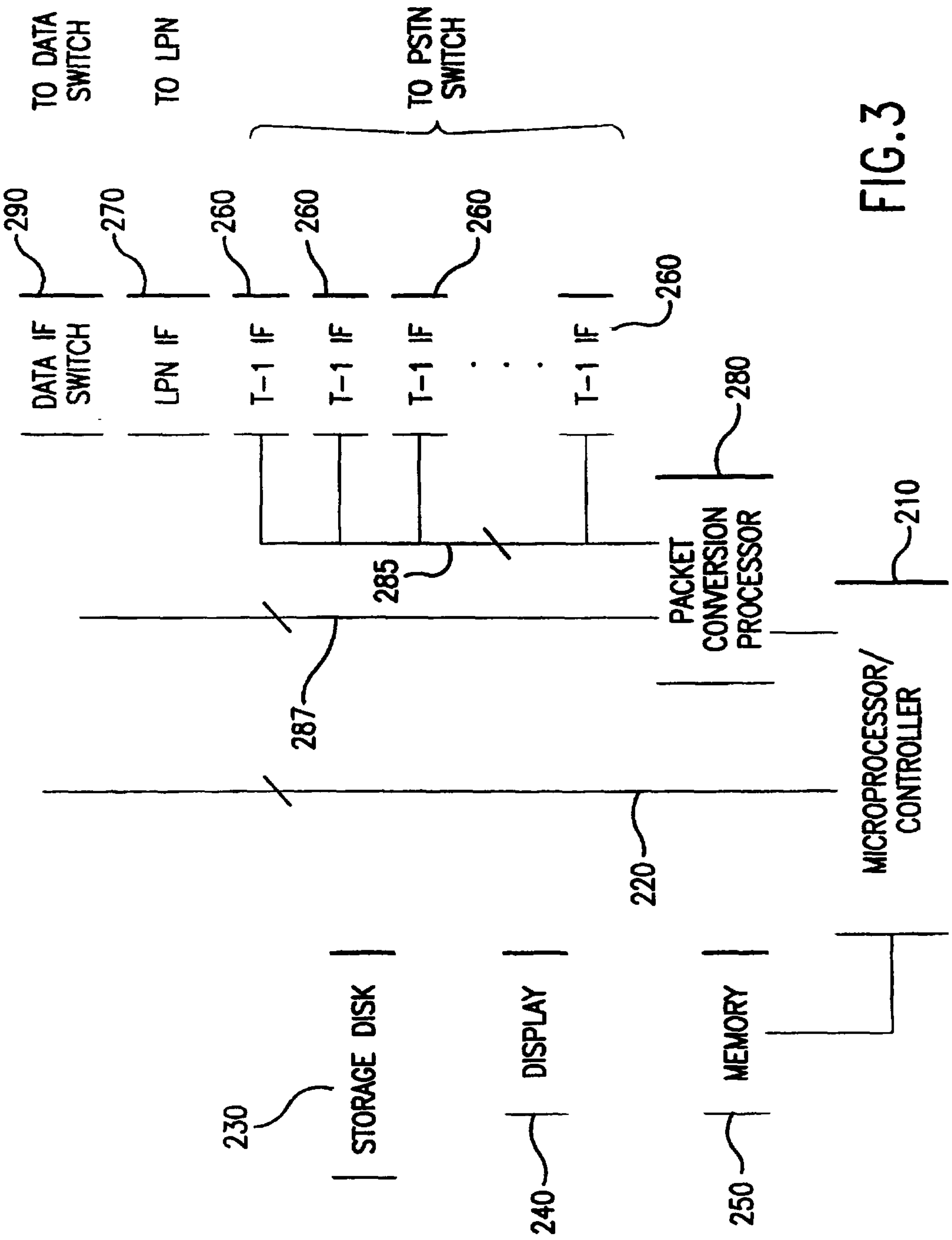

FIG.3

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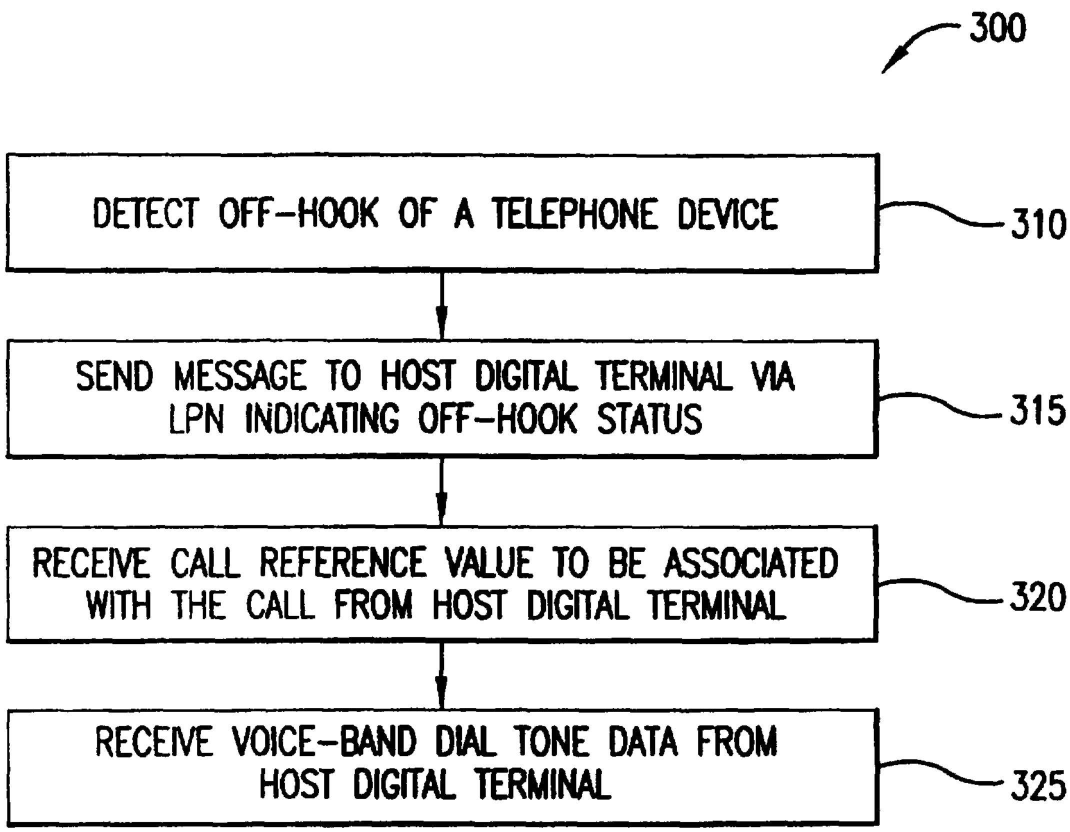
300
DETECT OFF-HOOK OF A TELEPHONE DEVICE
310
SEND MESSAGE TO HOST DIGITAL TERMINAL VIA LPN INDICATING OFF-HOOK STATUS
315
RECEIVE CALL REFERENCE VALUE TO BE ASSOCIATED WITH THE CALL FROM HOST DIGITAL TERMINAL
320
RECEIVE VOICE-BAND DIAL TONE DATA FROM HOST DIGITAL TERMINAL
325

FIG.4

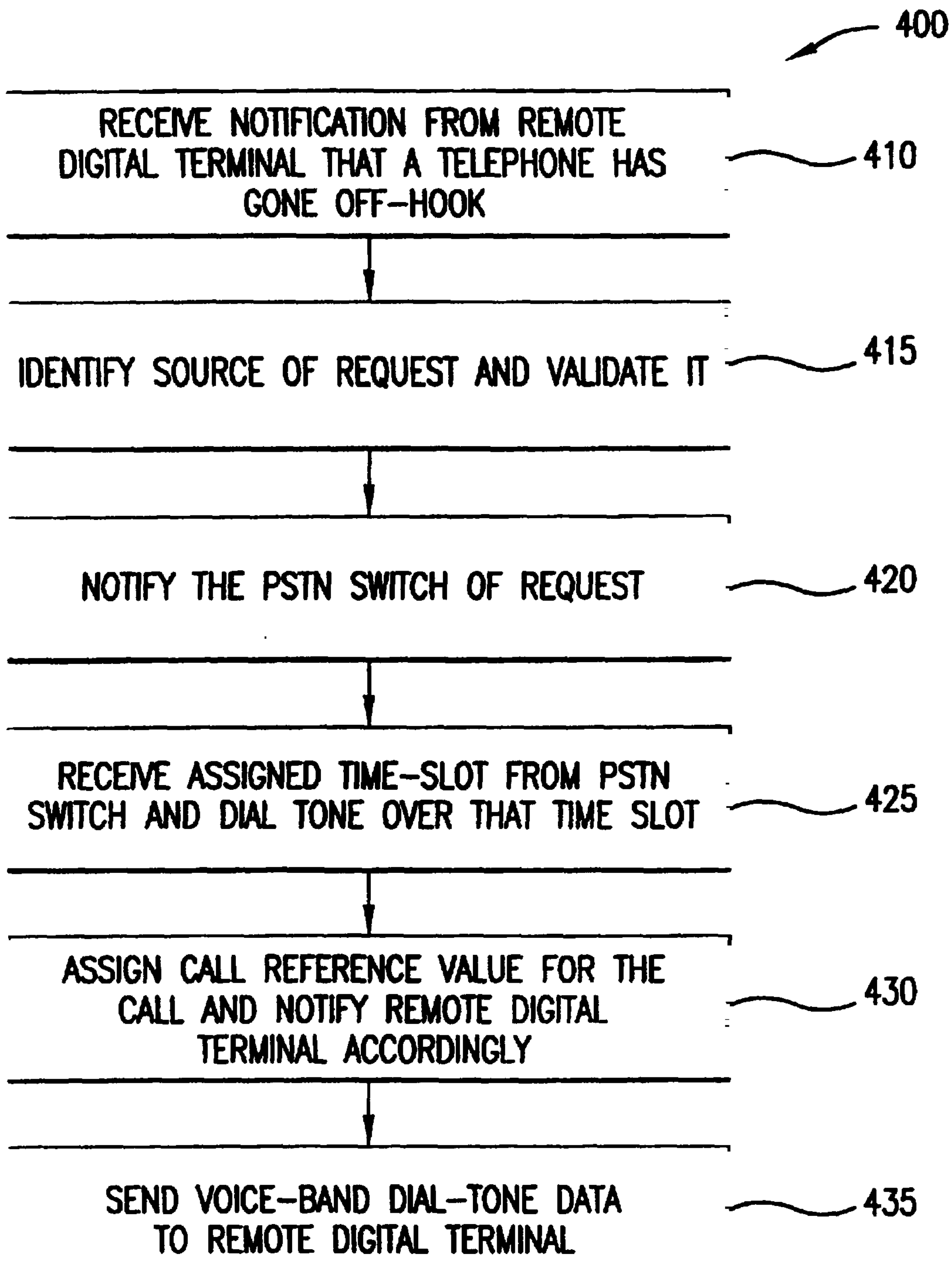

FIG.5

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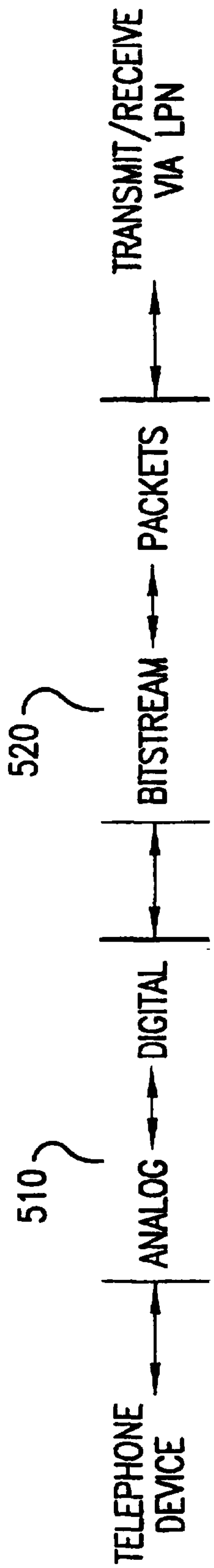

FIG.6

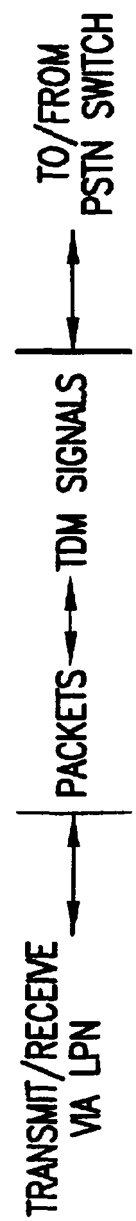

FIG.7

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SYSTEM AND METHOD FOR COMMUNICATING VOICE AND DATA OVER A LOCAL PACKET NETWORK

This application claims priority to U.S. Provisional Application, No.60/088,399, filed Jun. 8, 1998.

FIELD OF THE INVENTION

The present invention is directed to a system and method for providing voice and data services over an access network supporting a digital packet-based transport protocol, such as Digital Subscriber Line (DSL) technology.

BACKGROUND OF THE INVENTION

The Telecommunications Act of 1996 officially opened the local telecommunications market in the United States to competition. Prior to 1996, the Regional Bell Operating Companies (RBOCs) held monopolies on local telephone service within their regions. As a result of the Act, the RBOCs were designated Incumbent Local Exchange Carriers (ILECs) and companies competing with the ILECs are referred to as Competitive Local Exchange Carriers (CLECs) To date, states have registered hundreds of new and established companies as CLECs, and some are now offering competitive local service. CLECs offering local service can choose from two basic network strategies for providing service. A CLEC can purchase its own switching and transmission equipment and build a local telecommunications network alongside the ILEC network. CLECs following this strategy are referred to as facilities-based CLECs. Alternatively, the Telecommunications Act of 1996 made provisions for CLECs to rebrand and resell ILEC services purchased at a discount. CLECs following this strategy are referred to as resale CLECs. In some cases, a CLEC will pursue both strategies.

Although many CLECs initially pursued a resale network model, most are now exclusively focused on providing facilities-based local service. The resale model had initial appeal because it enabled a CLEC to quickly offer a broad array of services to both business and consumer customers in many geographic markets with little initial capital investment. However, the profit margins from resale proved inadequate as a viable long-term business strategy.

There are two major components to a local telecommunications network, the switching (or core) network and the access network. As an oversimplification, the switching network provides the service while the access network transports the service to the customer. For an ILEC, multimillion dollar digital switches located in every community in a geographic market constitute the switching network, while the thousands of pairs of copper wires that run from each ILEC central office (CO) to customer premises constitute the access network.

CLECs, however, demand local networks that are very different from those used by ILECs. The CLECs do not need, nor could they afford, to immediately build-out decentralized switching networks to service every potential subscriber in a market. Therefore, CLECs choose to centralize their switching systems, using one or two digital switches to service an entire market. However, while a centralized switching strategy is more efficient and less expensive, it presents a challenge to design an access network capable of serving a small, geographically dispersed customer base. An access network must accommodate the increased distance between the switching equipment and each customer site. As an added challenge, most CLECs intend to offer local telephone service and data services, such as high speed Internet access. Consequently, CLECs need access networks that are capable of delivering both voice and data services to avoid the complexity and expense of constructing separate voice and data access networks. To meet these challenges, most CLECs are turning to broadband access networks: access networks that enable a high bandwidth connection to be established between the CLECs centralized facility and remote customer sites. Transmission equipment is now available that can enable a CLEC to deliver voice and data services over several of these new broadband access networks.

Broadband access networks generally consist of two components. The first is called the backbone network that connects the centralized switching equipment of a CLEC to a centralized location within each community, such as the ILEC central office. The backbone portion of a CLEC broadband access network is usually a fiber optic network, such as one that conforms to the Synchronous Optical Network (SONET) standard. The second component is called the "last mile" network and is the connection from the community location to each customer site. There are a variety of broadband technologies available for the last mile portion, and the one selected for use by a CLEC can greatly affect the capital investment required to serve a community. The last mile broadband access technologies currently available for use in access networks by CLECs are Fiber-to-the-Building (FTTB), Hybrid Fiber/Coax (HFC), Wireless Local Loop (WLL) and Digital Subscriber Line (DSL).

Traditionally, telephone subscribers have been connected to the Public Switched Telephone Network (PSTN) through a last mile network that physically consists of a pair of copper wires running between a subscriber (home or office) and the "wire center" of the telephone company. The wire center typically serves thousands, or even tens of thousands, of subscribers in a neighborhood or community, and houses a "central office switch" that terminates each subscriber wire pair. The switch controls the telephone at each subscriber site, providing power, ringing, and audio signals in analog form over the wire pair. The switch also detects when a subscriber line goes "off hook," dials digits, etc., and in response routes and connects calls to other subscribers or to other switches in the PSTN. This in summary is known as the "Plain Old Telephone Service" (POTS) and is an analog technology (as opposed to digital technology).

Subscribers requiring more than a single telephone "line" can be served by installing the corresponding number of POTS circuits using multiple copper wire pairs. Alternatively, a "pair gain" system can be deployed, which multiplexes the signals for several telephone lines onto a single pair of wires. This is accomplished with special equipment at each end of the copper pair. The pair gain equipment converts a POTS analog signal into a digital format, usually at 64 kilobits per second. A digital connection is established over the copper pair, with sufficient bandwidth to carry all of the required bit streams. Time-division multiplexing is used to merge the bit streams. A typical pair gain arrangement is a "T-1" line, operating at 1.536 mega-bits per second (Mbps), allowing it to carry 24 individual 64 kilobit/second channels or streams. At a subscriber location, special equipment converts each of the 24 streams to and from a format of conventional POTS signals. At the wire center, similar equipment is required.

The T-1 technology is not always more economical than utilizing the existing infrastructure of the POTS system because T-1 service requires deployment of a significant amount of special equipment and infrastructure. Moreover,

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if a subscriber requires less than 24 lines of service, the T-1 solution is even less attractive because the equipment cost is spread over fewer lines. At the central office switch, a T-1 line is terminated by the special port that is dedicated to that subscriber, even if only a fraction of the 24 channels are used by that subscriber.

DSL is a high bandwidth technology that enables data to be transferred to and from individual subscriber locations at various speeds, currently ranging as high as 2 Mbps. Data is transferred over a DSL access portion of a local packet network (LPN) as "packets," and packets move over the LPN only when information is moving to or from the subscriber, and the line is in an idle condition otherwise. An LPN is a network that provides data connections among subscribers in a local service area with various connection types and data rates. Typically, an LPN consists of a plurality of DSL multiplexers and data switches. DSL equipment is designed to serve large numbers of subscribers, resulting in relatively low per-subscriber costs.

DSL technology stands out as being the most attractive to a CLEC in terms of up front capital investment. FTTB, HFC and WLL broadband access technologies each require the installation of significant infrastructure (fiber optic cables, coaxial cables, base stations and repeaters, etc.) which is not economical for a CLEC to service a decentralized small or medium-sized business customer base. DSL runs over existing copper last mile networks (local loops) of the ILECs and therefore does not require significant capital expenditures for deployment. Instead, the CLEC pays a monthly fee to the ILEC for each of the local loops that it uses. In addition, DSL has the correct capacity for serving residential through medium-size business markets.

In general, the DSL access portion of a local packet network does not carry voice and is not connected to the central office switch. However, some implementations of DSL can combine, on a single wire pair, both the digital signal carrying data packets, and a single POTS signal carrying analog voice. This permits a DSL subscriber to use the line for a telephone call while simultaneously transferring packet data. However, this approach is limited to a single POTS signal, and requires POTS compatible equipment to terminate the line at the wire center, in addition to the packet-oriented DSL equipment.

It is desirable to provide a system that enables facilities-based full service CLECs to transport local telephone service, including multiple voice call service, and data services to small and medium-sized business customers over an access network that supports a digital packet-based transport protocol, preferably over existing copper wire pair lines. It is further desirable that the system use the fewest number of local loops, and that a minimum amount of CLEC equipment be required at the ILEC wire center.

SUMMARY OF THE INVENTION

Briefly, the present invention is directed to a system and method for utilizing a local packet network (LPN) that supports a digital packet-based transport architecture, such as Digital Subscriber Line (DSL), to provide voice and optionally data services over a single local loop, such as a DSL, to a customer site. Multiple voice telephone calls as well as data services for a customer site are supported on a single DSL connected to that customer site.

At a customer site, a plurality of telephone devices (such as telephones, facsimile machines, modems and/or office telephone system ports) and data devices (such as those connected by a local area network) are interfaces to a local loop link connected to the LPN. Analog telephone signals (representing voice, facsimile signals, or modem signals) received from the plurality of telephone devices are converted to digital voice-band packets. Control signals representing off-hook, dial tone, call setup information, and other call control signals are converted to digital call control packets. The voice-band packets, call control packets and data packets (from the data devices at the customer site) are modulated for transmission via the local loop link over the LPN. In the reverse direction, modulated voice-band packets, data packets and call control packets received from the LPN destined for the customer site via the LPN on the local loop link are demodulated. The demodulated voice-band packets are converted to analog telephone signals for connection to appropriate ones of the plurality of telephone devices at the customer site. The demodulated data packets are coupled to the data devices (in the local area network) at the customer site. The demodulated call control packets are processed to control call setup and maintenance functions at the customer site.

At a control site within or connected to the LPN (such as a central switching facility), voice-band packets, call control packets and data packets from the customer site are received via the LPN. The voice-band packets received from the customer site via the LPN are converted to time-division multiplexed signals and are coupled to a public switched telephone network (PSTN) switch in assigned time slots. Data packets received from the customer site are coupled to a data switch for transfer to a data network (such as the Internet). The call control packets are processed to control call setup and maintenance functions at the control site. In the reverse direction, data packets destined for the customer site are received from the data switch and coupled to the LPN for transmission to the customer site. Time-division multiplexed voice signals received from the PSTN switch destined for the customer site are converted to voice-band packets and are coupled to the LPN for transmission to the customer site.

According to the present invention, a specialized apparatus called a remote digital terminal (RDT) is provided at the customer site and another specialized apparatus, called a host digital terminal (HDT) is provided at the central switching facility. Alternatively, the specialized functions of the HDT are integrated into a data switch in the LPN or into a PSTN switch. Similarly, the functions of the RDT can be integrated into a key telephone system/private branch exchange device or other equipment at the customer site The RDT and HDT transport digitized voice-band packets and data packets between each other via the LPN. The RDT converts the voice-band packets suitable for communication over the LPN to and from analog telephone signals suitable for use by attached telephone devices. Similarly, the HDT converts voice-band packets to and from a time-division multiplexed format suitable for communication via the PSTN switch.

The above and other objects and advantages of the present invention will become more readily apparent when reference is made to the following description, taken in conjunction with the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram of a telecommunication system employing the remote digital terminal and host control terminal of the system and method according to the present invention.

FIG. 2 is a block diagram of a remote digital terminal according to the present invention.

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FIG. 3 is a block diagram of a host digital terminal according to the present invention.

FIG. 4 is a flow chart depicting a call set-up procedure in the remote digital terminal.

FIG. 5 is a flow chart depicting a call-set up procedure in the host digital terminal.

FIG. 6 is a flow chart depicting processing of voice-band data in the remote digital terminal.

FIG. 7 is a flow chart depicting processing of voice-band data in the host digital terminal.

DETAILED DESCRIPTION OF THE INVENTION

FIG. 1 is a general diagram of a system **5** that enables a facilities-based full service CLEC to transport local telephone service and data services to small and medium-sized businesses. The present invention is described as being useful in utilizing a particular type of local loop link, called the DSL network. However, it should be understood that the teachings described herein are applicable to any access network supporting a digital packet-based transport protocol. DSL is only an example of such a protocol-access network technology.

The primary components of the system according to the present invention are a remote digital terminal (RDT) **100** and a host digital terminal (HDT) **200**. The RDT **100** resides at a customer (subscriber) site shown at reference numeral **7** and interfaces a plurality of telephone devices (TDs) **10** and, optionally, a local area network (LAN) **12** to a local loop link, such as a line supporting the Digital Subscriber Line (DSL) transport protocol. For simplicity, the local loop link is referred to as a DSL **24**. Other local loop links that may be suitable for use in conjunction with the present invention are wireless local loops, such as digital cellular local loops, and the like.

The DSL **24** is an access network of a local packet network (LPN) **60**. The LPN **60** comprises one or more data switches **62**, such as Asynchronous Transfer Mode (ATM) switches, and one or more DSL access multiplexers (DSLAMs) **64**. The data switches **62** consist of one or more processors controlled by software. The data switches **62** are connected to each other and to DSLAMs **64** preferably via optical links, such as synchronous optical network (SONET) facilities **66**. In each ILEC central office (CO), there is a DSLAM **64** that controls the distribution (and collection) of signals to and from a plurality of DSLs **24**. The combination of DSLAMs **64** and data switches **62** make up the LPN **60**.

The LPN **60** provides data connections among subscribers in a local service area with various connection types and data rates. For example, the LPN might include DSL connections at rates ranging from 256 Kbps to 6 Mbps used by homes and small businesses, T3 connections at 45 Mbps used by large business and small Internet Service Provides (ISPs) and OC-3 connections for use by the largest businesses and ISPs.

Returning to the description of a customer site **7**, the TDs **10** may connect directly to the RDT **100** or to a key telephone system/private branch exchange (KTS/PBX) device **14** that is connected to the RDT **100**. Connections between the RDT **100** and the associated TDs are by standard analog telephone lines **25**, or alternatively by other standard telephony interfaces such as T-1, ISDN, etc. Connections between the RDT **100** and the KTS/PBX **14** is by way of a plurality of trunks **27**. The function of the RDT **100** is to allow voice traffic associated with one or more TDs **10** and data traffic from the LAN **12**, if any, to be converted to and from a format that can transit the LPN **60**. It should be understood that the voice-band traffic associated with the TDs **10** could be voice, modulated digital data from a modem, facsimile machine, and possibly certain call control signals (such as dial tone, busy signal, etc.). Data traffic is that traffic associated with the LAN **12** or other data-packet based devices at the customer site **7**.

As is well known in the art, DSL is a telecommunication technology that enables data to be transferred from individual subscriber locations at various speeds, currently as high as 2 Mbps using the existing twisted wire pair line infrastructure already in place in most industrialized parts of the United States and the world. That is, the DSL **24** is a standard twisted wire pair line that is used to transmit information that is formatted in accordance with the DSL transport protocol.

The HDT **200** resides at control site within or without the LPN **60**. For example, FIG. 1 shows the HDT residing at a CLEC switch facility **30** and interfaces a PSTN switch **32** and a data switch **34** to the fiber backbone of the LPN **60**. The PSTN switch **32** may route voice calls to the local PSTN **42** or to a long distance network. The data switch **34** may route data packets to and from a data network **50**, such as the Internet. The HDT **200** links via an optical fiber **26** or another facility connected to the LPN **60**. The CLEC **30** switching facility is also hereinafter referred to as a central switching facility. Both the data switch **34** in the central switching facility **30** and the data network **50** may route data directly to the LPN **60**, bypassing the HDT **200**. Similarly, a data switch **62** in the LPN **60** may directly route data to the data network **50**.

Alternatively, the functions of the HDT **200** may be incorporated into a control site at another location in the system. For example, the functions of the HDT may be incorporated into a data switch **62** of the LPN **60** rather than be performed by a separate unit. Software that carries out the functions of the HDT **200** (described hereinafter) may be provided in the data switch **62** to be executed by the processor(s) in the data switch **62**. The software in the data switch **62** can be enhanced to perform the functions of the HDT **200** and interface directly to the PSTN switch **32**, to the data switch **34** or to the data network **50**. Similarly, the software may be provided in a PSTN switch **32** to be executed by processors associated with the PSTN switch **32**.

Turning to FIG. 2, with continued reference to FIG. 1, the components of the RDT **100** will be described. In a preferred embodiment, the RDT **100** is embodied as circuit board housed in a suitable enclosure with a power supply.

Specifically, according to a preferred embodiment, the RDT **100** comprises a controller **110**, a DSL modem **120**, a static random access memory (SRAM) **130** for buffering working data, a read only memory (ROM) **140** that stores a software program for the controller **110**, a plurality of subscriber line interface circuits (SLICs) **150**, a plurality of coder/decoders (CODECs) **160**, and an Ethernet interface **170**.

The DSL modem **120** connects directly to the DSL **24**. The DSL modem performs the modulation and demodulation necessary to transport information via the DSL **24** into the LPN **60**. There are several modulation/demodulation formats that are known in the art for use over a DSL **24**. The DSL modem **120** also formats the modulated information into a suitable packet format, such as the asynchronous transfer mode (ATM) protocol for example, that is utilized by the equipment in the LPN **60** for the transport of information. Alternatively, if the local loop link is a wireless local loop link, the modem **120** would be a wireless modem capable of performing the modulation and demodulation necessary for transporting information via a wireless link. In addition, a transceiver (not shown) would be connected to the model **120** to wirelessly transmit and receive the modulated information.

The controller **110** is connected to the DSL modem **120**, the SRAM **130**, ROM **140**, Ethernet interface **170** and a time division multiplex (TDM) bus **180**. Furthermore, the controller is connected to each of the SLICs **150** by a control bus **155** to enable the controller **110** to detect when a TD **10** goes off-hook and to command an SLIC **150** to ring a TD **10**. Each SLIC **150** is connected to a TD **10** or to a PBX/KTS **14** by a standard analog telephone line. The SLIC **150** provides the precise voltages and currents required to interface to telephone devices, such as standard telephone sets, facsimile machines, etc. The CODEC **160** is a coder/decoder that converts analog telephone signals (voice and other in-band telephone signals) to digital bit streams, and converts digital bit streams to analog telephone signals. The controller **110** is connected to each of the CODECs **160** by the TDM bus **180**. The number of TDs **10** services by the RDT **100** will determine the number of SLICs **150** and CODECs **160** required. However, as will be explained hereinafter, the number of telephone devices that may be in use at any one time depends on the bandwidth of the local loop link. The Ethernet interface **170** is a standard network interface circuit device that interfaces digital data between an attached PC or LAN 12.

The controller **110** is preferably a microprocessor that operates in accordance with software stored in the ROM **140**. The operation of the controller **110** may be updated or modified by employing a reprogrammable non-volatile ROM, such as a "flash" memory, that is well known in the art. The controller **110**, under control of the software program stored in the ROM **140**, performs two major functions: call set-up control and voice-band data conversion.

As an alternative to the implementation shown in FIG. **2**, the RDT **100** may be implemented using a PC with plug-in cards that provide the necessary interfaces (DSL, telephone, and Ethernet). Still another alternative is to provide a server computer that provides the call control functionality to a plurality of multimedia client PCs each having plug-in telephony and sound cards so that each PC can support a telephone call. Yet another alternative is to implement the functions of the controller **110** by a digital signal processor (DSP) or an application specific integrated circuit (ASIC).

Furthermore, rather than providing a plurality of individual SLICs **150**, a single subscriber line interface unit capable of coupling a plurality of signals to and from a plurality of telephone devices may be employed. Similarly, a single voice conversion device having the processing capability to perform multiple conversions may be used in place of the plurality of separate CODECs **160**.

The function of the RDT **100** is to allow traffic associated with one or more telephone calls to be converted to and from a form that can transit the LPN **60**. More specifically, the functions of the RDT **100** include interfacing to a local loop link, such as a DSL (via an integrated or external DSL modem); converting voice-band packets to and from conventional analog telephone signals; converting data packets to and from a format suitable for transport via the LPN; processing call control functions (ringing, on-hook, and off-hook functions) to generate and detect call control packets sent or received via the LPN; providing an electrical

interface to conventional telephone equipment; managing the sharing/allocation of the bandwidth on the local loop link with other (non-telephony) functions; and providing remote management and maintenance functions.

Turning to FIG. **3**, in conjunction with FIG. **1**, the HDT **200** will be described. The HDT **200** performs functions complementary to those performed by the RDT **100**. A single HDT **200** can support communication with a plurality of RDTs **100** that are connected to the LPN **60**.

In the preferred embodiment, the HDT **200** is implemented in a conventional computer system, with specialized software controlling a set of interface electronics. Specifically, the HDT **200** shown in FIG. **3** comprises a microprocessor-based controller **210**, a bus, such as a Peripheral Connection Interface (PCI) bus **220**, a storage disk **230**, a display **240**, and a memory **250**, such as SRAM. The HDT **200** interfaces to the PSTN switch **32** by a PSTN interface (IF) device, such as T-1 IFs **260**, preferably communicating via the Bellcore GR-303 signaling interface. That is, the T-1 IFs **260** connect to T-1 lines, which are in turn connected to the PSTN switch **32**. The HDT **200** interfaces to an optical link connected to the LPN **60** by an LPN IF device **270**, such as an OC-3 optical link utilizing the Asynchronous Transfer Mode (ATM). Similarly, a data switch IF device **290** is provided to interface with the data switch **34**. PC-cards implementing the functions of the LPN IF device **270** and data switch IF **290** are plugged into the PCI bus **220**.

The LPN IF device **270** connects to the LPN **60** via an optical fiber (FIG. **1**) and provides a data transfer rate of 155 Mbps in each direction. ATM cells are sent and received over the link, wherein each cell contains address information, including source and destination information, as well as the data to be transferred. Alternatively, the LPN IF device **270** operates at other rates, such as OC-12 at 622 Mbps, Fast Ethernet (100 Mbps) or Gigabit Ethernet (1000 Mbps.) However, the preferred embodiment is the ATM-155 standard.

The T-1 IFs **260** are coupled to a packet conversion processor **280** by a TDM bus **285**. Similarly, the data switch IF **290** and LPN IF **270** are coupled to the packet conversion processor **280** by a TDM bus **287**. The function of the packet conversion processor **280** is to convert information between different formats used by the devices connected to it. For example, the packet conversion processor **280** converts packets received from the LPN via the LPN IF **270** to a time-divisional multiplexed format for coupling to the T-1 IFs **260**. Similarly, the packet conversion processor **280** converts packets received from the LPN IF **270** to a suitable format for coupling to the data switch IF **290**. The packet conversion processor **280** performs these conversions in reverse as is appropriate. The packet conversion processor **280** is preferably implemented by an application-specific processor, and its operation is supervised by the microprocessor/controller **210** for call control functions, call setup, system errors and other matters. In some configurations, the function of the packet conversion processor **280** may be included within the functionality of the microprocessor **210**.

The T-1 facilities connected to the PSTN switch **32** are logically divided into "time-slots" using time-division multiplexing. Each T-1 line carries 24 time-slots or channels, with each channel carrying a single digitized voice conversation. One or more of the 24 channels is designated a control channel and carries signaling information between the HDT **200** and the attached PSTN switch **32**. Use of the control channel is described in more detail hereinafter.

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Preferably, the HDT **200** is designed to have a modular architecture that is easily scaleable. The HDT **200** can cost-effectively support as few as 100 end user lines and 24 trunks to the voice switch **32**, and up to as many as 6,000 lines and 2,000 trunks. Additional HDTs can be added to handle as many as 100,000 lines or more. The HDT **200** is preferably designed to provide "carrier-class" availability, including redundant and "hot-swappable" components. It is preferably a NEBS level 3 compliant rack-mounted system designed to reside in a central office environment and can be engineered to have N+1 redundancy.

A software program stored in the memory **250** allows the microprocessor **210** to perform the control functions analogous to the call set-up and functions performed by the RDT **100**, and to support communication with multiple RDTs **100** simultaneously. As mentioned above, software to carry out the functions of the HDT **200** may be incorporated directly into the PSTN switch **32** or into the data switch **62** such that a separate "box" to carry out the functions of the HDT **200** would not be needed.

The operations of the RDT **100** and HDT **200** are described with reference to FIGS. **4–7**, together with FIGS. 1–3. In order to transport voice and data services via the DSL access portion through the LPN **60**, the RDT **100** and HDT **200** employ a compatible digital signaling and information transfer protocol. There are many such protocols well known in the art of telecommunications, and it is likely that many new protocols will be created that may be useful in connection with the system and method of the present invention. The ATM protocol is the example of a suitable protocol. The ATM protocol is a format that divides a bandwidth into a plurality of cells each of which may contain voice-band packets, data packets, call control packets, etc.

The local loop link supports in each direction the transport of voice-band packets representing analog voice-band telephone signals, data packets associated with data devices (computers in a LAN) and call control packets. A voice-band packet includes an identifier (source and destination) and the voice-band packet information. A call control packet includes a control flag (identifying it as a control message), a control message (off-hook, busy, etc.) and an identifier associated with the call to which the control message applies. If the ATM protocol is used, each packet occupies an ATM cell. Upon receiving a packet, the RDT **100** and HDT **200** first detect the type of packet received (by a control flag or other identifying information) in order to determine whether the packet represents call control functions, voice-band information or data information. The RDT **100** converts analog telephone signals to and from a suitable digital packet format in order to communicate with the attached analog TDs **10**. In addition, if communication with data devices at the customer site is supported and required, the RDT **100** converts data packets from the data devices to and from a suitable packet format for communication via the LPN. Similarly, the HDT **200** converts voice-band information and data packets between different types of digital formats in order to conduct communications via the PSTN switch **32** and the data switch **34**.

Turning to FIG. **4**, with reference to FIG. **2**, a call-setup procedure **300** in the RDT **100** is described. This procedure occurs when a call is initiated by one of the TDs **10** connected to the RDT **100**. In step **310**, an SLIC **150** detects that a connected TD **10** is off-hook, and a corresponding signal is coupled to the controller **110**. The controller **110**, in step **315**, generates a control message (formatted into a call control packet) indicating the off-hook status and requesting a dial-tone, that is transmitted to the HDT **200** via the LPN **60**. The controller **110**, in doing this, first determines whether there is available bandwidth on the DSL **24** based on the number of other voice calls currently being maintained by the RDT **100**. If there is available bandwidth, then the message is sent to the HDT **200**; otherwise, a "busy" or other signal indicating unavailability is sent to that telephone device.

Next, in step **320**, the RDT **100** receives a call control packet from the HDT **200** that includes a call reference value to be associated with that call. In step **325**, a call control packet or voice-band packet (depending on the system implementation) representing a voice-band dial-tone is received from the HDT **200** that is converted to a digital bit stream by the RDT **100**, converted to an analog dial-tone signal by the CODEC **160** and connected to the TD **10** by the SLIC **150**.

FIG. **5** illustrates the complementary call set-up procedure **400** in the HDT **200**, again, in the case when a call is initiated by a telephone device connected to the RDT **100**. The use of the ATM protocol is referred to in the following description as an example of a suitable networking technology used by the LPN. Reference is also made to FIG. **3** in connection with this description. In step **410**, the HDT **200** receives the call control packet containing a control message from the RDT **100** indicating that a telephone device has gone off-hook. In step **415**, the HDT **200** identifies the source of the call control message by looking at the address provided in the ATM cell containing the call control packet request. It is verified whether the request is valid, i.e., coming from a customer site **7** whose account is active. In step **420**, the HDT **200** notifies the PSTN switch **32** via the designated control channel over one of the T-1-IFs **260** of the request. In reply, the PSTN switch **32** assigns an available time-slot for the call on one of the T-1 facilities, and this time-slot information is received by the HDT **200** in step **425**, together with a dial-tone over that time-slot. Next, in step **430**, the HDT **200** assigns a call reference value for the call and communicates that value in a control message that is transmitted to the RDT **100**. Finally, in step **435**, the HDT **200** generates voice-band packet(s) representing a dial-tone for the call and transmits it to the RDT **100**.

Calls that are initiated by devices on the HDT side of the system are received by the HDT **200** when the PSTN switch **32** detects a match between a received telephone number and a table of telephone numbers assigned to a customer site within the supervisory control of the HDT **200**. In response, the HDT **200** transmits to the RDT **100** a call control packet containing a command for a ringing signal that is interpreted by the RDT **100** to generate an analog ringing signal for connection to the appropriate TD **10**. A call reference value is assigned, and voice-band packets are exchanged between the HDT **200** and RDT **100** in a manner analogous to that described for a call initiated at the RDT **100**. When the addressed telephony device at the RDT **100** goes off-hook, the RDT **100** generates a control packet indicating same, and the HDT **200**, upon receipt of that packet, notifies the PSTN switch **32** that the call has been answered. Thereafter, the operation of the RDT **100** and HDT **200** continues as explained below. The communication of a data packet to a data device at the customer site is initiated in an analogous manner when the data switch **34** receives a data packet destined for a data device at the customer site.

It should be understood that the "telephone numbers" associated with the TDs **10** connected to the RDT **100** are managed by the PSTN switch **32** located in the central switching facility **30**. When a person somewhere in the

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PSTN 42 dials a number, such as 404-555-1234, traditional PSTN technology routes that call to the PSTN switch 32, which in turn presents the call information, as described herein, to the HDT 200, for transmission to the RDT 100 and ultimately to the connected TD 10. Thus, the TDs 10 connected to the RDT 100 operate identically to telephone devices connected directly to the central switching facility 30, and can make and receive PSTN telephone calls in the traditional fashion.

Once a call between the RDT 100 and HDT 200 is set up, remaining communication will consist primarily of voice-band packets or data packets depending on the end devices communicating with each other until the call is terminated. In general, only when new calls are initiated, calls are terminated, or system errors occur will call control packets occur over the local loop link.

The operation of the RDT 100 once a call is set-up is described with reference to FIG. 6, in conjunction with FIGS. 1 and 2. There are two major conversion processes that occur at the RDT 100: analog-to-digital and digital-to-analog conversion shown at step 510, and digital bit stream-to-packet and packet-to-bit stream conversion at step 520. The analog-to-digital (and vice versa) conversion is performed by the CODECs 160 for the analog telephone signals. Some call control related telephone signals are generated by the controller 110 via the SLICs 150. The bit stream-to-packet (and vice versa) conversion is performed by the controller 110.

To explain further, the flow of signals form a TD 10 to the LPN 60 is first described. Analog telephone signals from an attached TD 10 are received and digitized by the attached CODEC 160, creating a digital bit stream representing the real-time analog telephone signals generated by the TD 10. The digital bit stream is placed on the TDM bus 180. In step 520, the digital bit stream is converted into digital packets called voice-band packets. Each voice-band packet contains a plurality of bytes of voice-band information, each representing a sample of the speech (or analog telephone signals) at a predetermined time interval, such as (125) microsec. A single voice-band packet contains a predetermined number of samples of voice information, such as 40, representing 5 msec. of speech, for example. More specifically, the controller 110 accepts 1 byte of digital data from the TDM bus 180 each 125 microsec., and buffers the information in the SRAM 130. When a full packet of voice information is accumulated in the SRAM 130, the controller 110 formats the packet with its associated call reference value and transfers it to the DSL modem 120, causing it to be modulated and transmitted via the LPN 60 to the HDT 200. The controller 110, under control of the software stored in the ROM 140, carries out this process for all calls that are active at the RDT 100. Similarly, when the controller 110 receives data packets via the Ethernet interface 170 from the LAN 12, it buffers them in the SRAM 130 for modulation by the DSL modem 120 and transmission via the LPN 60. Depending on the availability of bandwidth on the DSL 24, the controller 110 may buffer the digital data from the LAN 12 until sufficient bandwidth becomes available, giving prior to the voice-band information.

The controller 110 caries out this process for each of the TDs 10 which is active at any given time, sequencing through each CODEC 160, accumulating a voice-band information for each call, and passing it to the DSL modem 120 for transmission when it is filled. Construction of the packets is skewed by the controller 110 such that all packets are not ready for transmission simultaneously; rather, a packet for a first call is completed, and then the packet for the next call is completed a fraction of a second later, allowing the packet for the first call to be transmitted by the DSL modem 120 so that it is ready to accept the packet for the next call when it is ready. Alternatively, the voice-band data for several calls may be multiplexed together into a single voice-band packet that contains identifiers to each call and data for each call. This alternative technique reduces any potential delay that may occur as a result of the packetizing process.

In the reverse direction, the DSL modem 120 receives modulated voice-band packets and data packets transmitted over the DSL 24 from the central switching facility 30. The DSL modem 120 demodulates the modulated voice-band packets and modulated data packets and couples them to the controller 110. The controller 110 receives the demodulated voice-band packets and data packets, and identifies voice-band packets by the call reference value contained therein. The controller 110 stages the voice-band packets in SRAM 130, queuing subsequent voice-band packets as they are received. Simultaneously, the controller 110 takes individual bytes of data from the SRAM 130 and places them onto the TDM bus 180 at the rate of 1 byte every 125 microsec., for connection to an appropriate one of the CODECs 160 associated with the addressed TD 10. The CODEC 160 converts the digital bit stream data to analog signals telephone which are coupled by the SLIC 150 to the TD 10.

With respect to the received data packets, when the controller 110 detects the reception from the DSL modem 120 of a data packet not associated with telephony functions, it directs it to the Ethernet interface 170 where it is coupled (according to an associated address) to the appropriate data device in the LAN 12.

The RDT 100 sets the priority for utilization of the bandwidth of the local loop link between voice traffic and data traffic. For example, the controller 110 may be programmed to assign priority of use of the local loop link to voice traffic over data traffic. In this case, bandwidth over the DSL connection will be used for voice calls when they are active, but will be available for data traffic when some of the TDs 10 are idle. Typically, the bandwidth of a DSL is as high as 2 Mbps. The controller 110 may designate a portion of that bandwidth for voice traffic, for all or adjustable time periods during a day. Once the data traffic maximum is reached, no further data traffic would be permitted. These traffic parameters are programmable in the RDT 100, and if necessary, can be adjusted in real-time to accommodate sudden bandwidth allocation needs.

With reference to FIG. 7, in conjunction with FIGS. 1 and 3, the operation of the HDT 200 in processing ongoing calls is described. Voice-band packets and data packets sent by the RDT 100 via the LPN 60 are received by the LPN IF 270. The packet conversion processor 280 converts the voice-band packets to time-division multiplexed (TDM) signals in a process similar to that in the RDT 100, and couples the TDM signals via a T-1 IF 260 to the PSTN switch 32 in an assigned time-slot. Data packets received from the RDT 100 are (reformatted if required by the packet conversion processor 280 and) coupled to the data switch 34 by the data switch IF 290 under control of the microprocessor 210.

In the reverse direction, the HDT 200 receives TDM signals for a given call from a T-1 IF 260 at a rate of one byte every 125 microsec. In a process similar to that in the RDT 100, the packet conversion processor 280 buffers the bytes of data of the TDM signals in the memory 250 to form voice-band packets, which are then dispatched to the RDT 100 via the LPN IF device 270. Data packets received from

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the data switch **34** are coupled to the LPN IF device **270** for transmission to the customer site.

The HDT **200** may also be involved in bandwidth usage control. For example, the microprocessor **210** may be programmed to continuously monitor the amount of bandwidth of the local loop link used by a given RDT **100** at customer site for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic. The microprocessor **210** may be programmed to limit data traffic to the customer site from the LPN such that a predetermined portion of the total bandwidth available to the customer site on the local loop link used by that RDT **100** remains available for voice traffic. Furthermore, the microprocessor **210** may be programmed to generate a message to be sent to a data switch **62** in the LPN **60** indicating the amount of bandwidth available for data so that the data switches **62** in the LPN **60** can abide by this rule in transmitting (or not transmitting) data traffic to the HDT **200** destined for a particular RDT **100** at a particular customer site.

In order to be competitive in the local market, CLECs must deliver service that is of the same quality as that which the customer is currently receiving from the ILEC. To achieve this, the system of the present invention digitizes voice conversations using standard "μ-law" encoding (64 Kbps) and need not use voice compression (though compression can be used if it is desired to reduce the bandwidth utilization through the LPN). In addition, to avoid the delays (latency) inherent in some packet networks, the system according to the preferred embodiment uses ATM signaling and is optimized throughout to limit buffer sizes and queues, insuring that voice-band data moves expeditiously between the RDT and HDT.

In summary, the present invention is directed to a method for communicating voice to and from at least one customer site over a local packet network (LPN) supporting a packet-based transport protocol. At a customer site, a plurality of telephone devices are interfaced to a local loop link, such as a DSL or wireless local loop, connected to the LPN. Analog telephone signals received from the plurality of telephone devices are converted to voice-band packets and the voice-band packets are modulated for transmission via the local loop link over the LPN. In the reverse direction, modulated voice-band packets received from the LPN on the local loop link are demodulated. The demodulated voice-band packets are converted to analog telephone signals for connection to appropriate ones of the plurality of telephone devices. In addition to the communication of voice, the method further supports the communication of data to and from data devices (which data devices may be part of a local area network) at the customer site. Data packets from the data devices are modulated and transmitted via the LPN. In the reverse direction, data packets received from the LPN are demodulated and coupled to the data devices (in the local area network) at the customer site.

At a control site within the LPN (such as a data switch in the LPN) or connected to the LPN (such as a HDT or a PSTN switch at a central switching facility), voice-band packets (and optionally data packets) from the customer site are received (via the LPN if the control site is external to the LPN). The voice-band packets received from the customer site via the LPN are converted to time-division multiplexed signals and are coupled to a public switched telephone network (PSTN) voice switch in assigned time slots. Data packets received from the customer site are (reformatted if necessary and) coupled via a data switch to a destination data network, such as data network. In the reverse direction, data packets destined for the customer site are received from a source data network via the data switch connected to the source data network, such as data network. Time-division multiplexed signals received from the PSTN switch destined for the customer site are converted to voice-band packets and are coupled to the LPN **60** for transmission to the customer site. Similarly, data packets received from the source data network via the data switch are coupled to the LPN for transmission to the customer site.

Similarly, the present invention is directed to a system for communicating voice and data over an LPN to and from at least one customer site (usually a plurality of customer sites) connected to the LPN via a local loop link, comprising:

a remote digital terminal at the customer site, the remote digital terminal interfacing a plurality of telephone devices at the customer site to the local loop link to transmit and receive voice via the LPN; and

a host digital terminal at a control site within or connected to the LPN that interfaces voice calls between the remote digital terminal and a public switched telephone network (PSTN) switch via the LPN.

The system may support eh further transport of data between data devices (which may be part of a local area network) at the customer site and a data network within the LPN or a data network connected to the host digital terminal. Suitable hardware and the associated control functions required in the remote digital terminal and host digital terminal for transporting voice traffic and data traffic to and from a customer site are described above.

The present invention is further directed to an apparatus for communicating voice and data between a local packet network (LPN) and equipment in a central switching facility having a public switched telephone network (PSTN) switch connected to the PSTN and a data switch connected to a data network. The apparatus comprises an LPN interface device connected to the LPN to interface voice-band packets and data packets to and from the LPN; a PSTN interface device connected to the PSTN switch to interface signals to and from the PSTN switch; a packet conversion processor coupled to the LPN interface device and to the PSTN interface device that converts voice-band packets between a format suitable for transit via the LPN and a time-division multiplexed signal format suitable for coupling to the PSTN switch; and a controller. The controller is coupled to the LPN interface device, to the packet conversion processor and to the PSTN interface device, and manages the interface of voice-band packets between the LPN and the PSTN switch.

The apparatus further supports the communication of data and comprises a data switch interface device connected to a data switch at the central switching facility and to the packet conversion processor. The packet conversion processor converts data packets between a format suitable for transit via the LPN and a format suitable for coupling to the data switch. The controller of the host digital terminal is connected to the data switch interface device and controls the communication of data packets received from the remote digital terminal by the LPN interface device to the data switch interface device and communication of data packets received from the data switch by the data switch interface device destined for the customer site to the LPN interface device for transmission via the LPN.

Furthermore, the present invention is directed to a method for communicating voice and data to and from a customer site over a local packet network (LPN), the customer site being connected to the LPN via a local loop link. The method comprises steps of:

receiving voice-band packets at a control site from the customer site via the LPN;

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converting the voice-band packets received from the LPN to time-division multiplexed signals;

coupling the time-division multiplex signals in assigned time-slots to a PSTN switch; and

converting time-division multiplexed signals received from the PSTN switch destined for the customer site to voice-band packets and coupling the voice-band packets to the LPN for transmission to the customer site.

The method further supports the communication of data by coupling data packets received at the control site from the customer site via the LPN to a data network within or without the LPN; and coupling data packets received from the data network destined for the customer site to the LPN for transmission to the customer site.

The above description is intended by way of example only and is not intended to limit the present invention in any way except as set forth in the following claims.

What is claimed is:

1. A system for communicating voice and data over a local packet network (LPN) to and from at least one customer site connected to the LPN via a local loop link, comprising:

a remote digital terminal at the customer site, the remote digital terminal interfacing a plurality of telephone devices at the customer site to the local loop link to transmit and receive voice via the LPN, wherein the remote digital terminal comprises:

a modem for connection to the local loop link to modulate information for transmission over the LPN and to demodulate information received from the LPN;

a plurality of subscriber line interface circuits each suitable for connection to a telephone device;

a plurality of coder/decoders each for connection to a subscriber line interface circuit, each coder/decoder capable of converting analog telephone signals received from an associated subscriber line interface circuit to digital bit streams and converting a digital bit stream to analog telephone signals to be supplied to the associated subscriber line interface circuit;

a network interface device for connection to a local area network of data devices at the customer site to couple data packets from the local area network for transmission over the LPN and to couple data packets received from the LPN to the local area network;

a controller coupled to the modem, to the coder/decoders, to the network interface device and to the subscriber line interface circuits, the controller coupling data packets from the network interface device to the modem for transmission on the local loop link to the LPN and coupling demodulated data packets received from the modem to the network interface device, and the controller managing the interface of voice-band packets between the LPN and the customer site by:

converting digital bit streams received from one or more coder/decoders to voice-band packets and coupling the voice-band packets to the modem for transmission on the local loop link to the LPN; and

converting demodulated voice-band packets received via the modem from the LPN to digital bit streams and coupling the digital bit streams to an appropriate one of the coder/decoders; and

a host digital terminal at a control site within or connected to the LPN that interfaces voice calls between the remote digital terminal and a public switched telephone network (PSTN) switch via the LPN.

2. The system of claim 1, wherein the local loop link supports a Digital Subscriber Line (DSL) and its associated transport protocols, and wherein the modem comprises a DSL modem.

3. The system of claim 1, wherein the local loop link supports a wireless transport protocol, and wherein the modem comprises a wireless modem.

4. The system of claim 1, wherein the controller of the remote digital terminal assigns priority of use of a bandwidth of the local loop link between voice traffic and data traffic.

5. The system of claim 4, wherein the controller of the remote digital terminal assigns priority of use of the bandwidth of the local loop link to voice traffic over data traffic.

6. The system of claim 1, wherein the host digital terminal comprises:

an LPN interface device for connection to the LPN to receive and transmit packets via the LPN;

at least one PSTN interface device for coupling signals to and receiving signals from the PSTN switch;

a packet conversion processor coupled to the LPN interface device and to the PSTN interface device, the packet conversion processor converting voice-band packets between a format suitable for transit via the LPN and a time-division multiplexed signal format suitable for coupling to the PSTN switch; and

a controller coupled to the PSTN interface device, to the packet conversion processor and to the LPN interface device, the controller managing the interface of voice-band packets between the LPN and PSTN switch.

7. The system of claim 6, wherein the host digital terminal further comprises a data switch interface device for interfacing data packets between the host digital terminal and a data switch connected to a data network, the data switch interface device being connected to the packet conversion processor, the packet conversion processor converts data packets between a format suitable for transit via the LPN and a format suitable for coupling to the data switch, the controller of the host digital terminal being connected to the data switch interface device and controlling the communication of data packets received from the remote digital terminal by the LPN interface device to the data switch interface device and communication of data packets received from the data switch by the data switch interface device destined for the customer site to the LPN interface device for transmission via the LPN to the remote digital terminal.

8. The system of claim 7, wherein the controller of the host digital terminal continuously monitors the amount of bandwidth used by the remote digital terminal for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic, and limits data traffic destined for the customer site such that a predetermined portion of the total bandwidth available to the customer site remains available for voice traffic.

9. The system of claim 6, wherein the controller of the host digital terminal continuously monitors the amount of bandwidth used by the remote digital terminal for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic, and wherein the controller of the host digital terminal sends a message to a data switch in the LPN indicating the amount of bandwidth available for data on the local loop link.

10. The system of claim 6, wherein the LPN interface device is an optical device implementing an asynchronous transfer mode (ATM) protocol.

11. The system of claim 6, wherein the PSTN interface device comprises a plurality of T-1 interfaces.

12. A method for communicating voice and data to and from at least one customer site over a local packet network (LPN) supporting a packet-based transport protocol, each

customer site comprising a plurality of telephone devices and at least one data device, the method comprising steps of:

at a customer site:

converting analog telephone signals received from the plurality of telephone devices to voice-band packets and modulating the voice-band packets for transmission over the LPN via a local loop link between the customer site and the LPN;

demodulating modulated voice-band packets received from the LPN on the local loop link;

converting the demodulated voice-band packets to analog telephone signals for connection to appropriate ones of the plurality of telephone devices;

modulating data packets received from the at least one data device at the customer site for transmission over the LPN via the local loop link; and

demodulating modulated data packets received from the LPN on the local loop link and coupling the data packets to the at least one data device at the customer site;

at a control site within or connected to the LPN:

receiving voice-band packets from the customer site via the LPN;

converting voice-band packets received from the customer site to time-division multiplexed signals and coupling the time-division multiplexed signals to a public switched telephone network (PSTN) switch; and

converting time-division multiplexed signals received from the PSTN switch destined for the customer site to voice-band packets and coupling the voice-band packets to the LPN for transmission to the customer site.

13. The method of claim 12, wherein

at the control site, further comprising steps of:

receiving modulated data packets from the customer site via the LPN and coupling the modulated data packets to a destination data network via a data switch; and

receiving data packets destined for the customer site from a source data network via the data switch and coupling the data packets received from the data switch to the LPN for transmission to the customer site

14. The method of claim 13, wherein at the customer site, further comprising the step of assigning priority of use of a bandwidth of the local loop link between voice traffic and data traffic.

15. The method of claim 14, wherein the step of assigning comprises assigning priority of use of the bandwidth of the local loop link to voice traffic over data traffic.

16. The method of claim 13, wherein at the control site, further comprising the steps of:

continuously monitoring the amount of bandwidth of the local loop link used by the customer site for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic; and

limiting data traffic to the customer site from the LPN such that a predetermined portion of the total bandwidth available to the customer site on the local loop link remains available for voice traffic.

17. The method of claim 13, wherein at the control site, further comprising the steps of:

continuously monitoring the amount of bandwidth of the local loop link used by the customer site for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic;

sending a message to the data switch in the LPN indicating the amount of bandwidth available for data traffic destined for the customer site.

18. The method of claim 12, wherein the functions specified for the control site are performed by a data switch that is part of the LPN.

19. The method of claim 12, wherein the functions specified for the control site are performed by the PSTN switch in a central switching facility.

20. The method of claim 12, and wherein the functions specified for the control site are performed by a host digital terminal that connects to the PSTN switch at a central switching facility.

21. The method of claim 12, wherein the local loop link supports a Digital Subscriber Line (DSL) and its associated transport protocols.

22. The method of claim 12, wherein the local loop link supports a wireless packet-based transport protocol.

23. An apparatus for communicating voice and data between a local packet network (LPN) and equipment in a central switching facility comprising a public switched telephone network (PSTN) switch connected to the PSTN, the apparatus comprising:

an LPN interface device connected to the LPN to interface voice-band packets to and from the LPN;

a PSTN interface device connected to the PSTN switch to interface signals to and from the PSTN switch;

a data switch interface device connected to a data switch at the central switching facility;

a packet conversion processor coupled to the LPN interface device, to the data switch interface device and to the PSTN interface device, the packet conversion processor converting voice-band packets between a format suitable for transit via the LPN and a time-division multiplexed signal format suitable for coupling to the PSTN switch, and the packet conversion processor converting data packets between a format suitable for transit via the LPN and a format suitable for coupling to the data switch; and

a controller coupled to the LPN interface device, to the packet conversion processor, to the data switch interface device and to the PSTN interface device, the controller managing the interface of voice-band packets between the LPN and the PSTN switch, the controller controlling the communication of data packets received from the remote digital terminal by the LPN interface device to the data switch interface device and communication of data packets received from the data switch by the data switch interface device destined for the customer site to the LPN interface device for transmission via the LPN, wherein the controller continuously monitors the amount of bandwidth of a local loop link used by a customer site for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic, and limits data traffic destined for the customer site from the LPN such that a predetermined portion of the total bandwidth available to the customer site on the local loop link remains available for voice traffic.

24. An apparatus for communicating voice between a local packet network (LPN) and equipment in a central switching facility comprising a public switched telephone network (PSTN) switch connected to the PSTN, the apparatus comprising:

an LPN interface device connected to the LPN to interface voice-band packets to and from the LPN;

a PSTN interface device connected to the PSTN switch to interface signals to and from the PSTN switch;

a packet conversion processor coupled to the LPN interface device and to the PSTN interface device, the packet conversion processor converting voice-band packets between a format suitable for transit via the LPN and a time-division multiplexed signal format suitable for coupling to the PSTN switch; and

a controller coupled to the LPN interface device, to the packet conversion processor and to the PSTN interface device, the controller managing the interface of voice-band packets between the LPN and the PSTN switch, wherein the controller continuously monitors the amount of bandwidth of a local loop link used by a customer site for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic, and generates a message that is sent via a data switch in the central switching facility to the LPN indicating the amount of bandwidth available on the local loop link for data traffic destined for the customer site

25 In a system for communicating voice and data to and from a customer site over a local packet network (LPN), the customer site being connected to the LPN via a local loop link, at a control site within or connected to the LPN, a method comprising steps of:

receiving voice-band packets at a control site from the customer site via the LPN;

converting the voice-band packets received from the LPN to time-division multiplexed signals;

coupling the time-division multiplex signals in assigned time-slots to a PSTN switch;

converting time-division multiplexed signals received from the PSTN switch destined for the customer site to voice-band packets and coupling the voice-band packets to the LPN for transmission to the customer site;

coupling data packets received at the control site from the customer site via the LPN to a data network within or without the LPN;

coupling data packets received from the data network destined for the customer site to the LPN for transmission to the customer site;

continuously monitoring the amount of bandwidth on the local loop link used by the customer site for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic; and

limiting data traffic destined for the customer site from the LPN such that a predetermined portion of the total bandwidth available to the customer site on the local loop link remains available for voice traffic.

26. In a system for communicating voice and data to and from a customer site over a local packet network (LPN), the customer site being connected to the LPN via a local loop link, at a control site within or connected to the LPN, a method comprising steps of:

receiving voice-band packets at a control site from the customer site via the LPN;

converting the voice-band packets received from the LPN to time-division multiplexed signals;

coupling the time-division multiplex signals in assigned time-slots to a PSTN switch;

converting time-division multiplexed signals received from the PSTN switch destined for the customer site to voice-band packets and coupling the voice-band packets to the LPN for transmission to the customer site;

continuously monitoring the amount of bandwidth on the local loop link used by the customer site for voice traffic to determine remaining bandwidth availability on the local loop link for data traffic; and

sending a message to the LPN indicating the amount of bandwidth available for data traffic destined for the customer site.

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[54] APPARATUS AND METHOD FOR COMMUNICATING VOICE AND DATA BETWEEN A CUSTOMER PREMISES AND A CENTRAL OFFICE

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[22] Filed: **Nov. 3, 1997**

Related U.S. Application Data

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[51] **Int. Cl.**[7] **H04B 1/38**; H04L 5/16

[52] **U.S. Cl.** **375/222**; 370/468; 370/494; 370/495

[58] **Field of Search** 375/222; 370/433, 370/468, 493, 494, 495; 379/88.13, 93.01, 93.09

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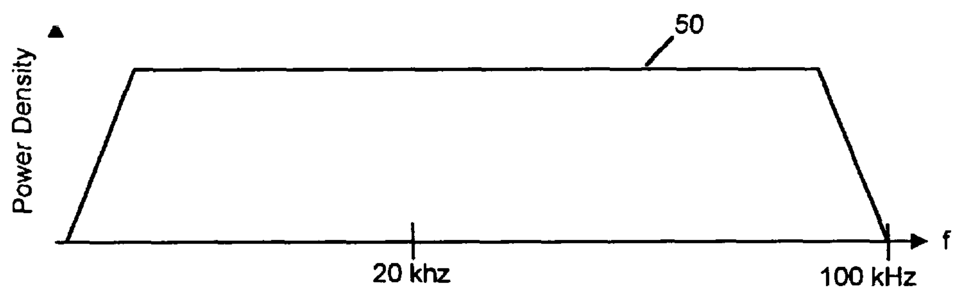
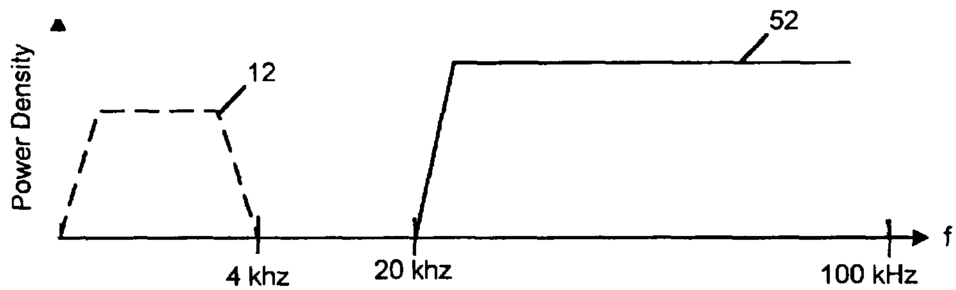
Attorney, Agent, or Firm—Thomas, Kayden, Horstemeyer & Risley, LLP

[57] **ABSTRACT**

A method and apparatus are provided for communicating data across a communication link, in a manner that senses and dynamically adapts to the simultaneous transmission of voice information across the local loop. In accordance with one aspect of the invention, a method is provided for dynamically communicating data over a local loop using a modem comprising the steps of transmitting data in a full-band transmission state, sensing a band-limiting condition, and adjusting the transmission of data from the full-band transmission state to a bandlimited transmission state, in response to the sensing step In accordance with the method, data may be transmitted by the modem across the local loop at the same time that voice information is communicated via telephone across the same local loop. A significant aspect of the present invention is the dynamic allocation of the data transmission bandwidth, whereby the invention senses a condition indicative of whether voice information is being communicated. If so, then the system shifts and/or narrows the data transmission bandwidth to allow for voice communications without interference from or with the data transmission. However, when no voice information is being communicated, the invention dynamically allocates the data transmission bandwidth to utilize at least a portion, if not all, of the frequency band otherwise used for communicating voice information.

54 Claims, 7 Drawing Sheets





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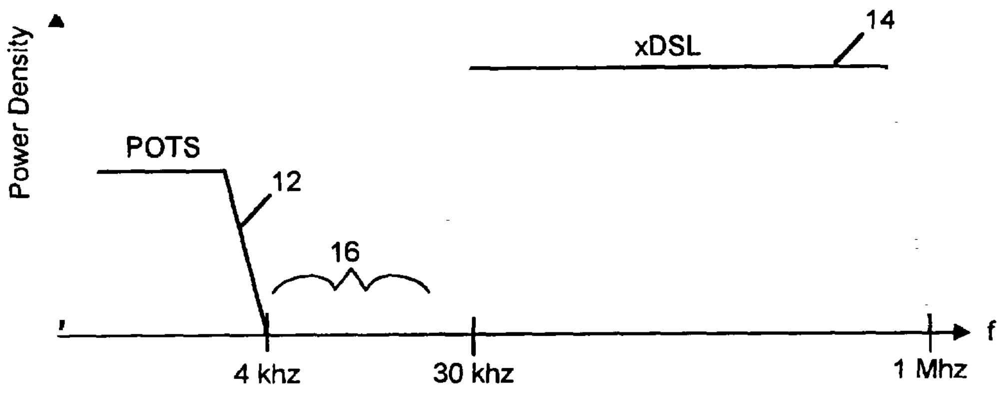

FIG. 1 (Prior Art)

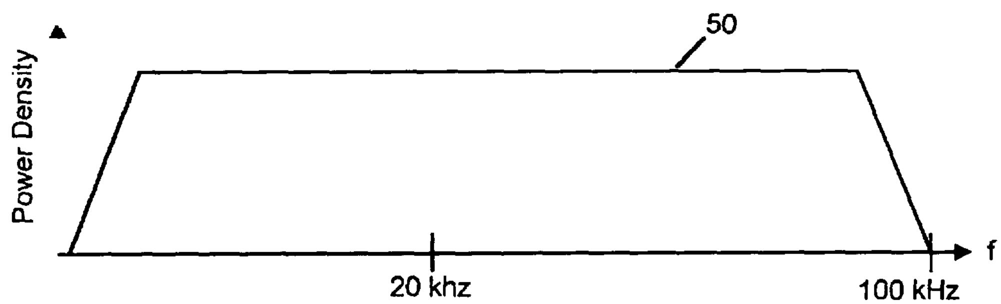

FIG. 3A

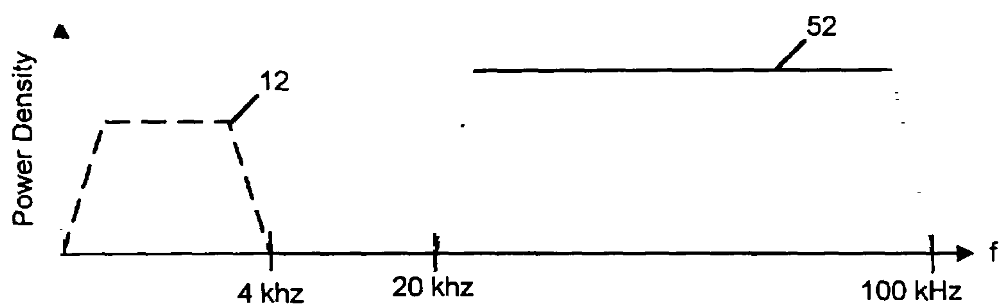

FIG. 3B

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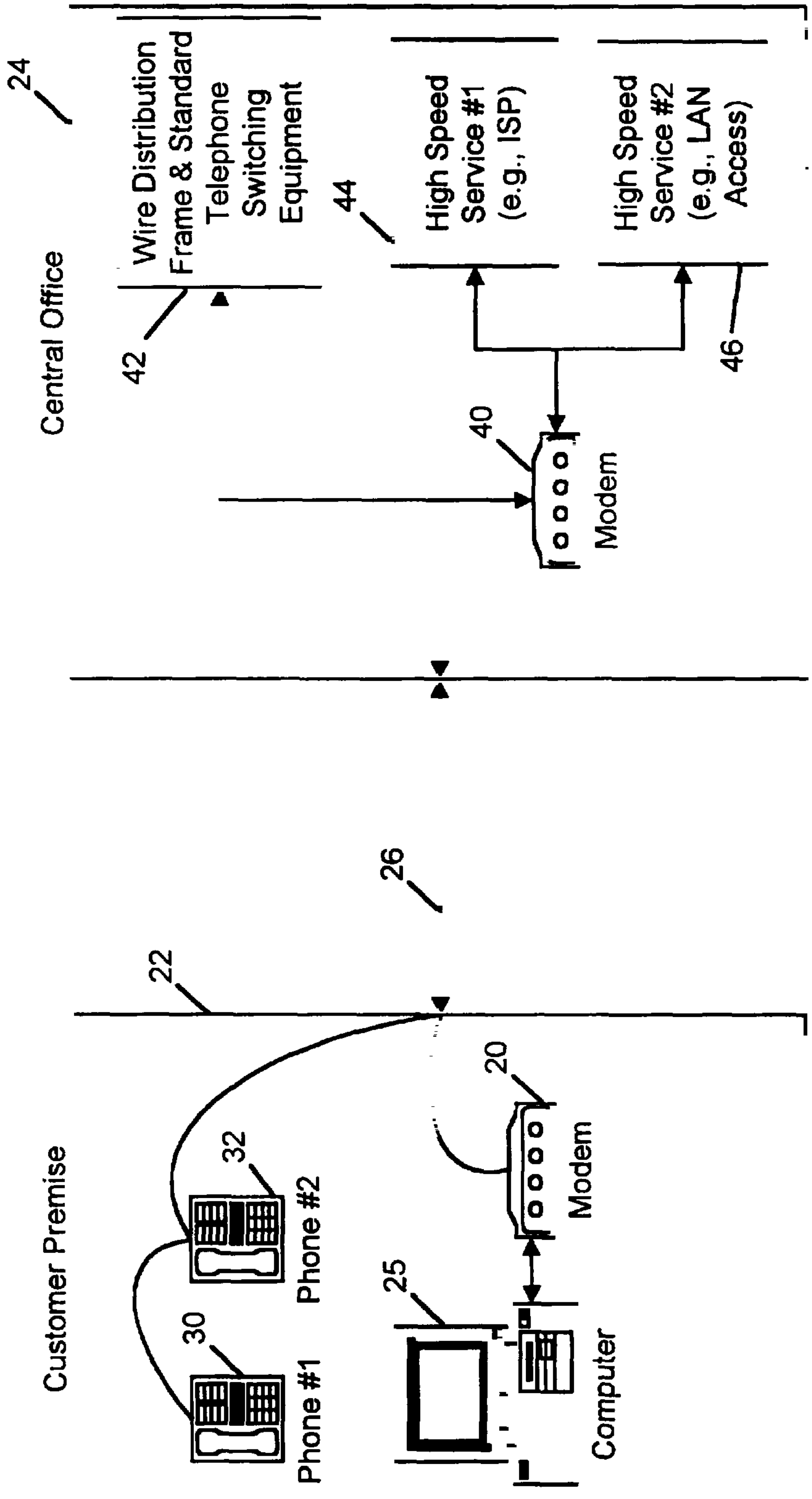

FIG. 2

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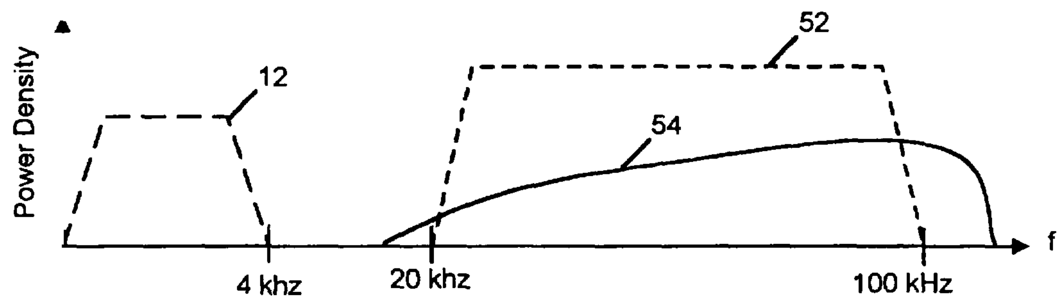
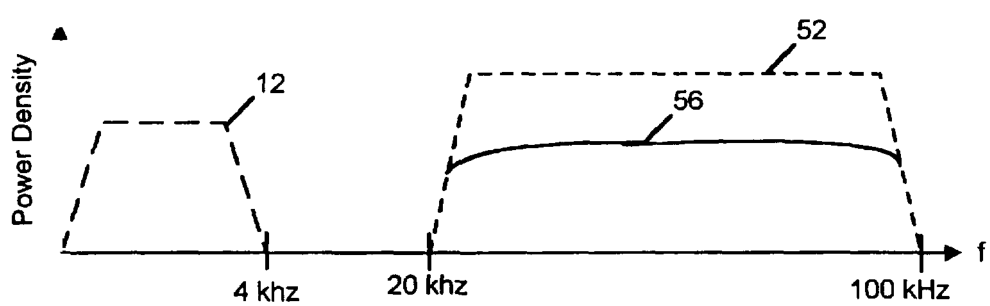


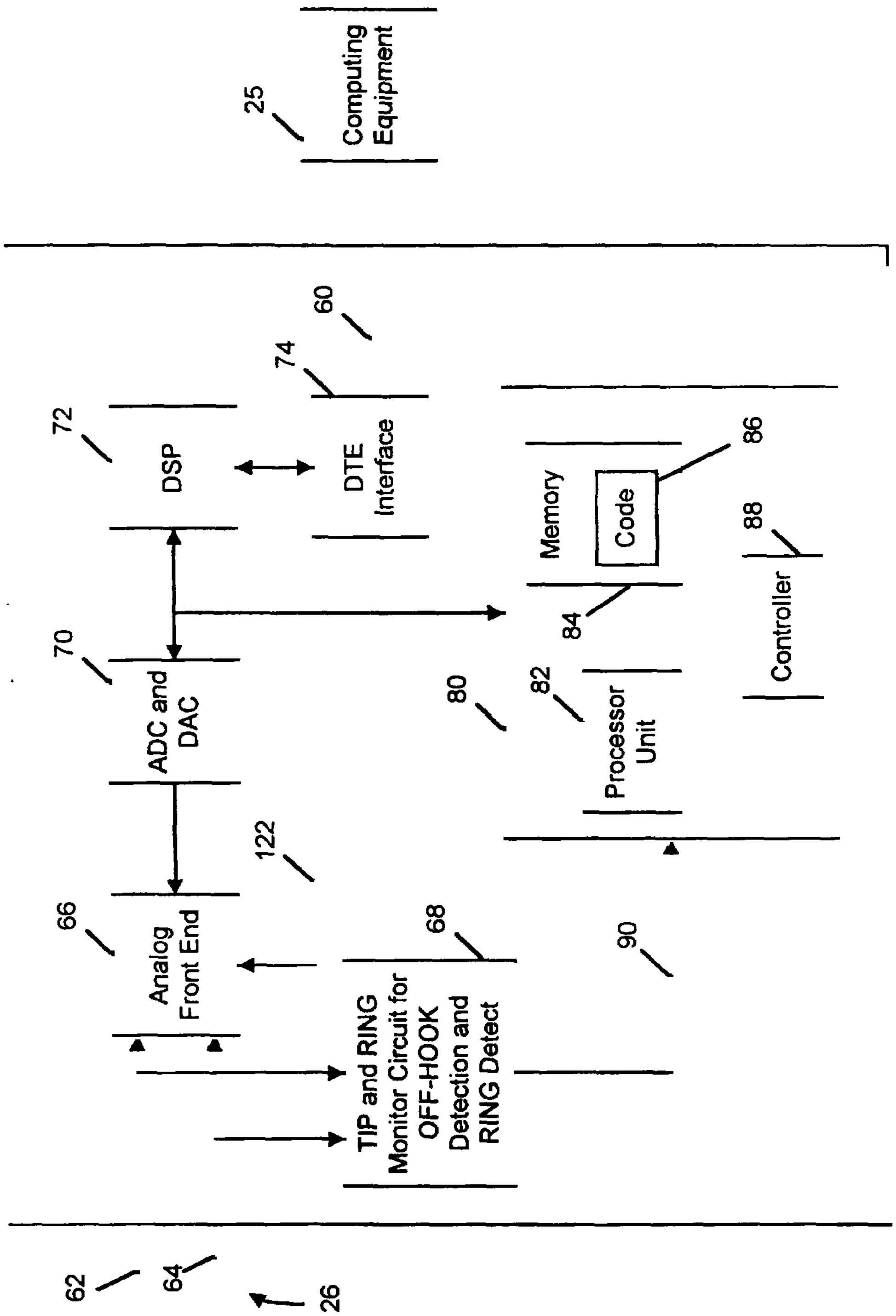
Power Density
52
12
54
4 khz
20 khz
100 kHz
f

FIG. 3C



Power Density
52
12
56
4 khz
20 khz
100 kHz
f

FIG. 3D



25
Computing Equipment
60
74
72
DSP
DTE Interface
Memory
Code
86
88
Controller
84
70
ADC and DAC
80
82
Processor Unit
122
66
Analog Front End
68
90
TIP and RING Monitor Circuit for OFF-HOOK Detection and RING Detect
62
64
26

FIG. 4

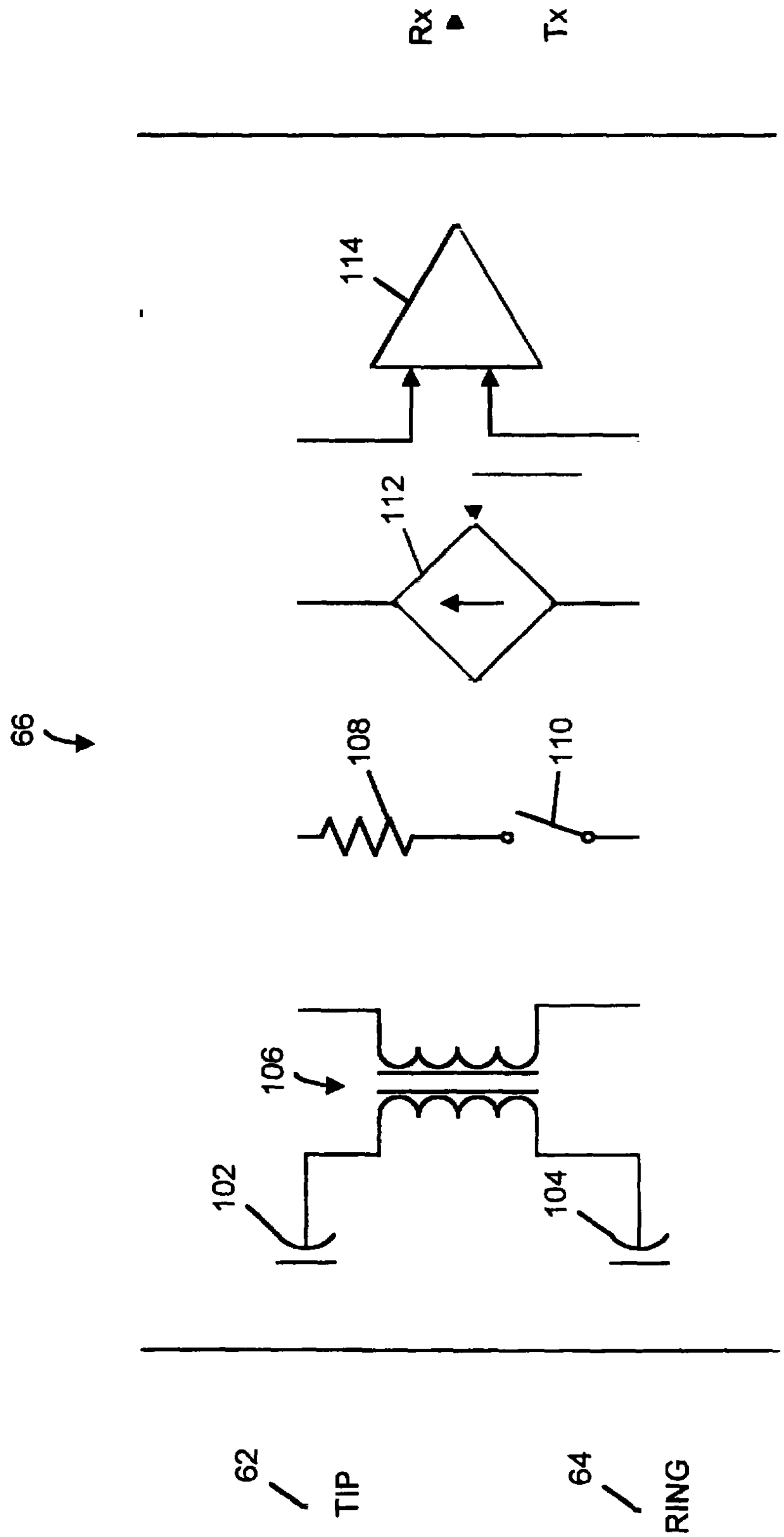
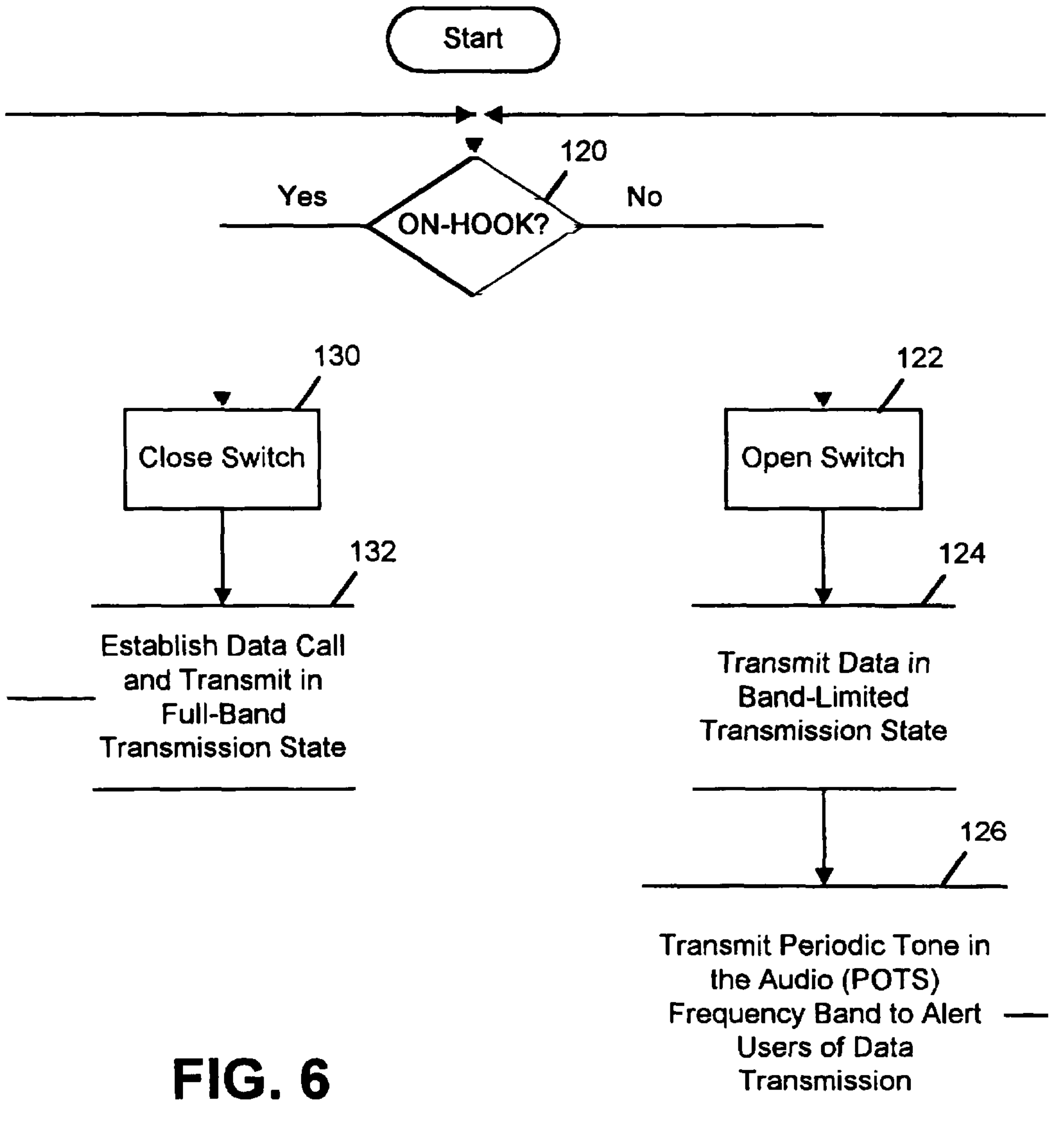
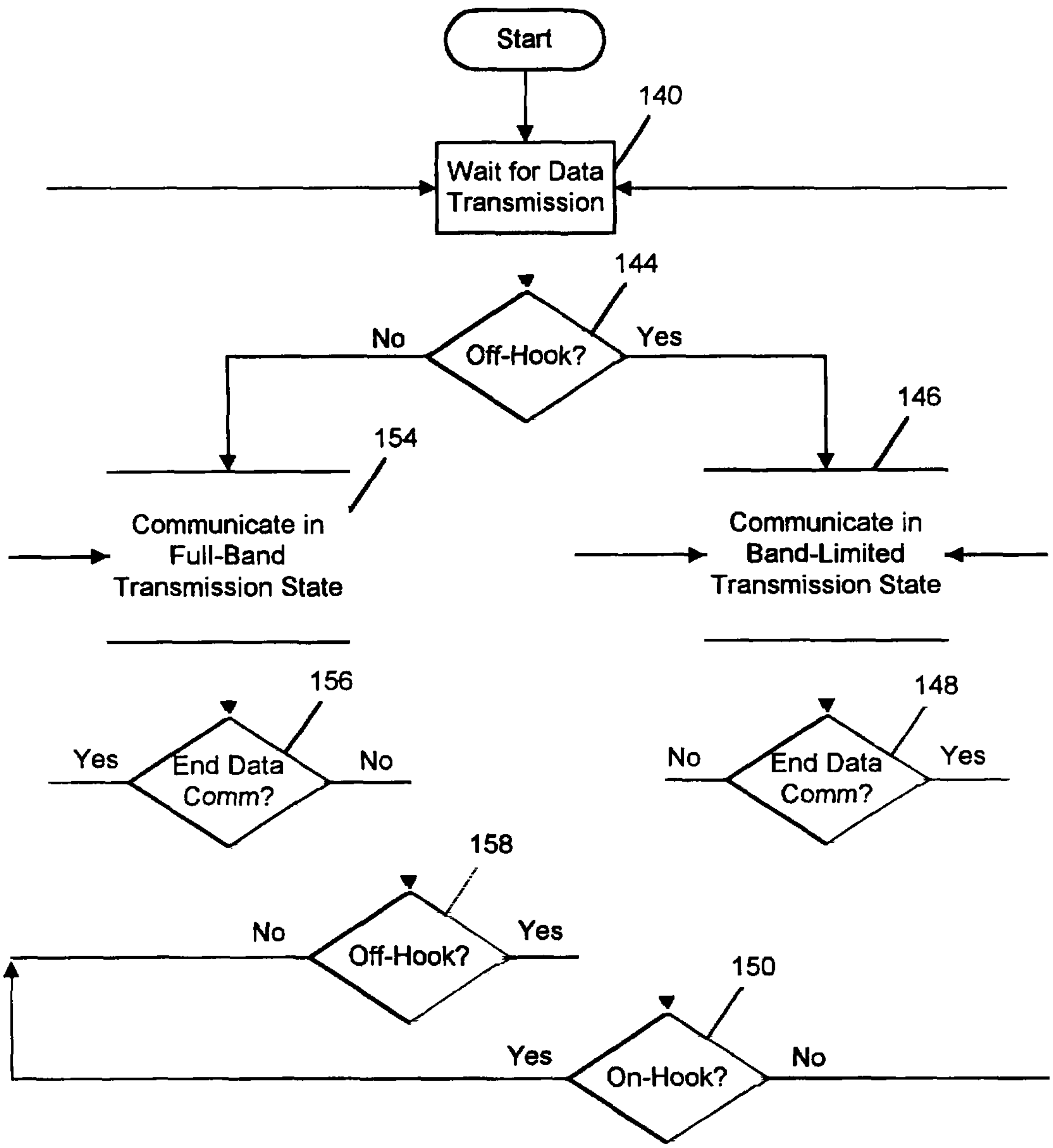


FIG. 5



Start
120
Yes
ON-HOOK?
No
130
Close Switch
132
Establish Data Call and Transmit in Full-Band Transmission State
122
Open Switch
124
Transmit Data in Band-Limited Transmission State
126
Transmit Periodic Tone in the Audio (POTS) Frequency Band to Alert Users of Data Transmission

FIG. 6



Start
140
Wait for Data Transmission
144
No
Off-Hook?
Yes
146
154
Communicate in Full-Band Transmission State
Communicate in Band-Limited Transmission State
156
Yes
End Data Comm?
No
148
No
End Data Comm?
Yes
158
No
Off-Hook?
Yes
150
Yes
On-Hook?
No

FIG. 7

APPARATUS AND METHOD FOR COMMUNICATING VOICE AND DATA BETWEEN A CUSTOMER PREMISES AND A CENTRAL OFFICE

CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of U.S. Provisional Patent Application Ser. No. 60/033,660, filed on Dec. 17, 1996, and entitled Digital Subscriber Loop Data Communications Method Enabling Simultaneous Data and POTS Without POTS Filters/Splitters or Special Premise Wiring.

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention generally relates to modems, and more particularly to high speed modems offering robust communication between a central office and a customer premises.

2. Discussion of the Related Art

High speed digital modems, such as Rate Adaptive Digital Subscriber Loop ("RADSL") modems are able to transfer data at high rates over the local loop, because they use frequencies which are significantly higher than the voice band frequencies used in Plain Old Telephone Service ("POTS"). By way of example, speech on a POTS system generally occurs in the frequency spectrum between about 0 Hz ("DC") and about 4 KHz, whereas RADSL modems use the frequency spectrum of between about 20 KHz to about 1 MHz. High speed digital modems generally include error detection circuitry which measures the errors which occur during communications. By making such measurements, they are then able to update their statistical knowledge of the wire pair which extends between the subscriber's location and the central office. Using that statistical knowledge, the modems can select optimal operating speeds. These modems were originally proposed when it was thought that services, such as video-on-demand, would be desirable.

As modem technology has developed, another need has arisen, in that the Internet has become a popular medium for both personal and work related use.

While the high speeds of RADSL modems seem to be quite desirable, their use of high frequencies mean that they also need to be protected from high frequency noise, such as cross-talk from adjacent channels or adjacent loops in the loop cable binder, as such noise causes them to downwardly adjust their operating speeds. In order to avoid certain types of noise, RADSL modems typically require the use of filters, called POTS filters, together with splitters for isolating Public Switched Telephone Network ("PSTN") equipment from the RADSL modems. Indeed, without POTS filters and POTS splitters, POTS signals directly interfere with the RADSL spectrum below about 20 kilohertz and the RADSL spectrum directly interferes with the POTS. POTS filters and POTS splitters reduce POTS signaling transients from interfering with RADSL data transmission. In addition, the use of the high RADSL bandwidth demands relatively high transmit power, which can cause distortions and dynamic range overload to POTS equipment.

Unfortunately, the manufacture and installation of POTS filters and splitters are expensive, and their use sometimes requires rewiring of the customer premises to ensure that all PSTN equipment is properly isolated from the RADSL modems and computing equipment. Consequently, it would be desirable to avoid the use of POTS splitters and filters, in

order to avoid the expense they impose (e.g., purchase cost and possible rewiring of customer premises).

Accordingly, their appears to be a need for a mass market modem which has data transfer rates greater than the 33.6 Kbps attainable by PSTN modems, yet under the rate that requires the addition of POTS filters, splitters, etc. to address noise and deleterious transmission line effects often encountered in high speed DSL modems.

Yet another problem which is manifest in increased Internet access and data communications is the increasingly limited availability to the customer phone line or local loop for its original purpose, i.e., voice communications. Of course, one solution is for a customer to purchase an additional phone line. This, however, imposes an additional cost on the customer. Moreover, unless the line is dedicated by the customer for a specific purpose (which is poor utilization), the second line may not always be available when needed.

Accordingly, there is a need to provide an improved modem that accommodates data transmissions, while simultaneously allowing traditional voice operation of a telephone attached to the same line at the customer premise. It is particularly desirable to have such a modem that does not require the use of costly POTS filters and splitters.

SUMMARY OF THE INVENTION

Certain objects, advantages and novel features of the invention will be set forth in part in the description that follows and in part will become apparent to those skilled in the art upon examination of the following or may be learned with the practice of the invention. The objects and advantages of the invention may be realized and obtained by means of the instrumentalities and combinations particularly pointed out in the appended claims.

To achieve the advantages and novel features, the present invention is generally directed to a method and apparatus for communicating data across a local loop, in a manner that senses and dynamically adapts to the simultaneous transmission of POTS (e.g., voice or PSTN modem) information across the local loop. In accordance with one aspect of the invention, a method is provided for dynamically communicating data over a local loop using a modem comprising the steps of transmitting data in a full-band transmission state, sensing a band-limiting condition, and adjusting the transmission of data from the full-band transmission state to a band-limited transmission state, in response to the sensing step. The step of sensing a band-limiting condition includes both the detection of the onset of a condition indicating that the method should enter the band-limited transmission state, as well as the detection of the cessation of that condition, indicating that the method should enter the full-band transmission state from the band-limited transmission state.

In accordance with the method of the present invention, data may be transmitted by the modem across the local loop at the same time that POTS (e.g. voice or PSTN modem data) information is communicated across the same local loop. A significant aspect of the present invention is the dynamic allocation of the data transmission bandwidth, whereby the invention senses a condition indicative of whether POTS information is being communicated. If so, then the system shifts and/or narrows the data transmission bandwidth to allow for voice communications without interference from or with the data transmission. However, when no POTS information is being communicated, the invention dynamically allocates the data transmission bandwidth to utilize at least a portion, if not all, of the frequency band otherwise used for communicating voice information.

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In accordance with the preferred embodiment, the method senses an off-hook condition of a telephone handset of a telephone electrically connected to the local loop. In use, a local loop extending between a customer premises and a central office branches, at the customer premise, to support multiple connections to the local loop. In this regard, the various branches or connections are typically routed throughout a customer premises to phone jacks, such as RJ-11 jacks. Multiple telephones may be plugged directly into these jacks for voice communication across the local loop. Similarly, a modem constructed in accordance with the present invention may be plugged directly into one of these jacks. The off-hook condition is preferably sensed by detecting either a change in impedance in the telephone line, or alternatively, a drop in line voltage across the telephone line.

In accordance with one embodiment of the invention, the full-band transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of less than about 15–20 kilohertz (and preferably less than 4 kilohertz). In the band-limited transmission state, the transmission frequency bandwidth has a lower frequency boundary of greater than 4 kilohertz. The significance of these values, for purposes of the invention, is that when no voice information is being communicated across the local loop, the transmission frequency bandwidth invades that frequency band generally dedicated to the transmission of voice information (i.e., the 0–4 kilohertz POTS frequency band). When, however, the invention senses that POTS information is being communicated across the local loop, or that there is a demand for the POTS band (e.g. telephone off-hook, ring, etc.), then the embodiment shifts the lower boundary of the transmission frequency bandwidth above the generally 4 kilohertz upper limit of the voice band. Preferably, the lower boundary will be shifted upwardly to approximately 20 kilohertz, to allow sufficient separation between the voice and data transmission frequency bands to that no interference between the two is realized, either by voice information corrupting data, or data transmission being heard in the voice band as noise.

For purposes of the preferred embodiment of the present invention, the precise value of the upper boundary of the transmission frequency bandwidth is not so significant, as it is the dynamic adjustment of the lower boundary and/or the reduced power in POTS mode, that realizes the inventive step. However, it will be appreciated that the upper boundary will generally be greater than 40 kilohertz, in order to define a meaningful transmission frequency bandwidth for data transmission. Indeed, in the preferred embodiment, the upper frequency boundary is approximately 80 kilohertz. It is believed that this frequency is low enough that transmissions may be effectively implemented without the need for POTS filters or POTS splitters, and therefore significantly reducing the cost of implementing the inventive system. Signal-to-noise ratio is high to permit reasonable data throughput without excessive power incident on attached POTS devices. Also, premises wiring and subscriber loop stubs do not cause substantive nulls in the frequency response. It will be further appreciated that shifting of the upper frequency boundary is not relevant to the present invention. That is, the upper boundary may be shifted in conjunction with the shifting of the lower frequency boundary, or alternatively, the upper frequency boundary may remain substantially fixed.

It will be further appreciated that—depending upon loading, line conditions, and other factors—the spectral shape of the band-limited xDSL transmission may be varied to minimize noise, intermodulation products, or other interference within the POTS frequency band. More particularly, it is generally understood that the power density of xDSL transmissions is generally greater than that of POTS transmissions. Merely shifting the xDSL transmission into the band-limited transmission state with a lower cut-off frequency of approximately 20 kHz may not always provide a wide enough guard band to prevent interference with the POTS band. Line loading, line conditions, and other factors (which differ among local loops) factor into this determination. Intermodulation products are another source of noise that often is present within the POTS band. When such noise is present within the POTS band, the band-limited transmission state may be further configured by reducing the power-density of the xDSL transmission. Another, related solution may be to uniquely shape the spectral curve for xDSL transmissions. This, for example, may be done by tapering the lower frequency portion of the curve (i.e., that portion near the approximately 15–20 kHz frequency).

In accordance with another aspect of the preferred embodiment, a modem is provided for communicating data across a local loop. The modem includes an input/output signal line that is electrically connected with the local loop (e.g., plugged into an RJ-11 phone jack). The modem also includes a processor unit that is adapted for operation in one of two states: a full-band transmission state and a band-limited transmission state. The full-band transmission state is defined by a lower frequency boundary at a value below approximately 15–20 kilohertz and an upper frequency boundary generally greater than 40 kilohertz (as discussed above). The band-limited state is defined by a lower frequency boundary greater than 4 kilohertz and an upper frequency boundary greater than 40 kilohertz (which may or may not be the same as the upper frequency boundary for the full-band transmission state). The modem further includes a sensor or other sensing means for sensing that the local loop is in POTS mode (e.g., transmitting POTS information, or preparing to transmit POTS information), and the data signal power and bandwidth are adaptively altered to provide data without out interfering with the POTS transmission. Upon sensing the band-limiting condition, such as an off-hook condition, the controller causes the processor unit to upwardly shift the lower frequency boundary of the transmission frequency band and operate in the band-limited, or reduced-power, state. Likewise, upon sensing no band-limiting condition (or a cessation in the band-limiting condition), the controller causes the processor unit to downwardly shift the lower frequency boundary of the transmission frequency band, and operate in the full-band transmission state, to maximize data throughput.

In accordance with yet a further aspect of the present invention, a method is provided for simultaneously communicating both voice and data between a customer premises and a central office across a local loop. In accordance with this aspect of the invention, the method comprises the steps of: (1) transmitting data between the customer premises and the central office in a first frequency band, wherein the first frequency band is defined by an upper frequency boundary and a lower frequency boundary; (2) allocating a second frequency band for transmitting voice information between the customer premises and the central office; (3) sensing a band-limiting condition; and (4) dynamically shifting the lower frequency boundary of the first frequency band in response to the sensed band-limiting condition. In accordance with the invention, the lower frequency boundary of the first frequency band shifted to at least partially overlap the second frequency band when no band-limiting condition exists. The lower frequency boundary of the first frequency

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band is further shifted to avoid overlapping with any portion of the second frequency band when the band-limiting condition exists.

In accordance with yet a further aspect of the invention, a modem is provided for communicating across a communication link capable of single-use transmissions and multiple-use transmissions. The term single-use transmissions is used to generally connote that a single transmission or communication is occurring across the link. For example, a single PSTN voice call, or a single data communication transmission. The term multiple-use transmissions is used to generally imply that multiple transmissions or communications are occurring simultaneously. For example, the simultaneous transmission of a data communication and a PSTN voice call. The modem constructed in accordance with this aspect of the invention includes an input/output signal line in communication with the communication link. It further includes a processor unit adapted for operation in one of at least two states, a full-band transmission state and a band-limited state, wherein the full-band transmission state occurs when single-use transmissions are occurring across the transmission link, and the band-limited transmission state occurs when multiple-use transmissions are occurring across the communication link.

It will be appreciated that, in accordance with a broad inventive aspect, the present invention operates by adjusting transmit power between a band-limited transmission state and a full-band transmission state. Generally (but not necessarily always), the full-band transmission state occurs when the communication link is operating in a single-use transmission mode, while the band-limited transmission state generally occurs when the communication link is operating in a multiple-use transmission mode. In accordance with this broad concept of the invention, substantial transmission energy is transmitted by the modem in or near the POTS frequency band, when the modem is transmitting in the full-band state. Conversely, very little (ideally zero) energy is transmitted by the modem in or near the POTS frequency band, when the modem is transmitting in the band-limited state. This allows for simultaneous POTS transmissions (e.g., voice, PSTN modem, etc.) in the POTS frequency band, and band-limited modem transmissions.

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 an illustration of the frequency spectrum of a dual frequency band communications system of the prior art, depicting the POTS transmission frequency band and the xDSL transmission frequency band;

FIG. 2 is a block diagram illustrating the primary components in a system utilizing the present invention;

FIG. 3A is a frequency spectrum illustrating the full-band transmission frequency band of the present invention;

FIG. 3B is a frequency spectrum illustrating the band-limited transmission frequency band of the present invention;

FIG. 3C is a frequency spectrum illustrating a band-limited transmission frequency band of an alternative embodiment of the present invention, having a uniquely shaped xDSL transmission band;

FIG. 3D is a frequency spectrum illustrating a band-limited transmission frequency band of an alternative embodiment of the present invention, having a reduced power xDSL transmission band;

FIG. 4 is a block diagram illustrating the primary components of a modem constructed in accordance with the present invention;

FIG. 5 is a circuit diagram illustrating the analog front end component of the modem block diagram of FIG. 4;

FIG. 6 is a software flowchart depicting the operation of the functional operation of the analog front end element, illustrated in FIG. 5; and

FIG. 7 is a software flowchart illustrating the top-level operation of a system constructed in accordance with the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT OF THE INVENTION

Having summarized the invention, 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.

Turning now to the drawings, FIG. 1 is a diagram illustrating frequency band communications, as is known in the prior art. The term frequency band communications is used to indicate communication of information within a certain defined, frequency band. As is known in the prior art, plain old telephone system (POTS) communications are transmitted in the frequency band 12 defined between about 0 (DC) and about 4 kHz. A second transmission frequency band 14 is defined at a higher frequency level than the POTS frequency band 12, and is used in the transmission of digital subscriber line (DSL) communications. A guard dead band 16 is typically provided to separate the two transmission frequency bands 12 and 14. The DSL transmission frequency band 14 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 14 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.

As will be discussed in more detail below, the present invention provides an upper transmission band having an upper frequency boundary that is much lower than the 1 MHz frequency boundary often encountered in xDSL transmissions. Indeed, the upper frequency boundary of the present invention is defined in a range that is readily supported by, or compatible with, transmission systems (and attached POTS-type equipment) presently in place between a customer premises and a central office, without the need for extraneous devices such as POTS filters and POTS splitters. In this regard, reference is made to FIG. 2, which is a top level diagram illustrating the principal hardware components of a system utilizing the present invention. In accordance with one aspect of the invention, a modem 20 is provided for achieving efficient data communications between a customer premises 22 and a central office 24

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across a local loop 26, by dynamically allocating a transmission frequency bandwidth and/or power 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 and/or power spectral density (PSD) 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. When, however, there is a demand for voice transmissions, then the present invention reallocates the transmission frequency band and PSD for the data communications so that there is no overlap or interference with the POTS transmission frequency band 12, and so that there is not significant interference to POTS-type attached equipment.

In keeping within the description of FIG. 2, the customer premises 22 may be a single-family household having a single phone line 26 for communicating between the customer premises 22 and a central office 24. Within the house or customer premises 22, multiple connections branch off of the local loop 26 and are terminated at phone jacks (such as RJ-11) located in various rooms of the household. In this way, multiple telephones 30 and 32 may be plugged in and supported from the same phone line 26. In the same way, a personal computer may be disposed in communication with the local loop 26 by way of a modem 20.

Presently, unless a user purchases an additional phone line, or a more costly communication service, such as xDSL, simultaneous transmissions of voice and data to different locations are not possible. As a result, one person in a household may have the local loop 26 tied up with data communications (such as Internet communications), while another person at the same household is awaiting the use of the local loop 26 for voice communication. In accordance with the present invention, and as will be discussed in more detail below, this shortcoming is overcome.

In keeping with the description of FIG. 2, a companion modem 40, that is compatible with the modem 20, is provided at the central office 24. As is known, other equipment, such as wire distribution frame and standard telephone switching equipment 42 may also be in communication with the local loop 26. Since the configuration and operation of such equipment is known in the prior art and does not effect or impact the present invention, it will not be discussed herein. FIG. 2 also illustrates a variety of services that may be connected at the central office 24 to the mode 40, constructed in accordance with the present invention. These services may include a high speed ISP service 44, a high speed LAN access service 46, etc. Again, since the provision and operation of such services are generally understood and are further not necessary in order to describe the operation of the present invention, they will not be described herein.

Turning now to FIGS. 3A and 3B, the dynamic allocation and deallocation of the data transmission frequency band is illustrated. Specifically, FIG. 3A illustrates the data transmission frequency band 50 in a full-band transmission frequency state, while FIG. 3B illustrates a data transmission frequency band 52 in a band-limited (POTS compatible) transmission frequency state. As illustrated in FIG. 3A, the full-band transmission frequency band 50 extends from approximately 0 Hz (DC) to approximately 100 KHz. In contrast, in FIG. 3B the data transmission frequency band 52 extends from approximately 20 KHz to approximately 100 KHz. In accordance with an important aspect of the preferred embodiment, a modem 20 constructed in accordance with the invention senses the need to dynamically allocate or deallocate a portion of the transmission frequency band in order to accommodate voice communications within the 0 to 4 KHz POTS frequency band 12. As will be described further herein, the present invention may sense this demand for voice transmissions (or band-limiting condition) by sensing an OFF-HOOK condition of a telephone 30, 32, (see FIG. 2) connected to the local loop 26. Alternatively, this band-limiting condition may be detected by an impedance change on the local loop 26.

For phone compatibility, in addition to detecting RING and OFF-HOOK conditions, the system may also be configured to detect voice conversation. Upon voice detection, the system may increase transmit power as it shifts into the band-limited transmission state, to increase data rate dynamically, so long as the voice band SNR is about 30 to 40 dB. When silence is once again detected (for a predetermined amount of time), the system will again reduce the transmit power for good idle channel perception.

Unlike typical xDSL communications, where the data transmission frequency band is often 1 MHz in width, the data transmission frequency band of the present invention is much less than that. This permits relatively high-speed data communication without the addition of expensive equipment, such as POTS splitters and POTS filters. Importantly, this addresses a market need from consumers that do not wish to incur, or cannot afford, the additional expenses normally incurred with purchasing an xDSL communication service. An important aspect of the present invention is its ability to sense when voice-band communications are not occurring, or otherwise when a band-limiting condition is not present, and expand the transmission frequency band into the frequency band otherwise reserved for POTS transmissions, and/or increase transmit power to increase the data rate. As can be seen from the illustrations in FIGS. 3A and 3B, expanding the transmission frequency band from a 20 kHz cutoff (FIG. 3B) to approximately DC (FIG. 3A) realizes a 25 percent increase in bandwidth (i.e., from 80 kHz to 100 kHz), and thus realize a significant improvement in performance.

FIGS. 3C and 3D illustrate alternative embodiments of the present invention. In short, FIGS. 3C and 3D illustrate a spectrally-shaped transmission curve and an adaptive power transmission curve, respectively. As illustrated in FIG. 3B, under normal operating condition, the power density of the xDSL transmission band is greater than that of the POTS transmission band. However, there may be instances when the guard band 16 is not large enough to sufficiently separate the xDSL transmission band 52 from the POTS frequency band 12. As a result, xDSL transmissions may be evident in the POTS frequency band 12 as noise (audible static). The reasons this may occur are varied, and include factors such as telephone set sensitivity and non-linearities. Intermodulation products may also be manifest within the POTS transmission band 12 as noise.

It will be appreciated that, consistent with the concepts and teachings of the present invention, various adaptations of the band-limited transmission state may be implemented to minimize or eliminate noise in the POTS transmission band 12. One solution is to further increase the size of the guard band 16, thereby increasing the frequency separation between the POTS transmission band 12 and the xDSL transmission band 52. Another solution is to adaptively reduce the transmit power of the xDSL transmission band. This solution is illustrated in FIG. 3D, wherein the normal power spectrum 52 is illustrated in dashed line and the reduced power spectrum 56 is superimposed in solid line. Reducing the transmit power in this way reduces the amount of noise that is manifest within the POTS frequency band. The specific amount of power reduction may vary among customer premises, based upon the attached equipment.

Yet another solution is to more particularly define the spectral shape of the transmission band. This solution is illustrated in FIG. 3C. As shown, the power spectrum of the xDSL transmission band 54 may be asymmetrically shaped to provide a greater taper on the lower frequency end of the curve. This taper, ensures sufficient attenuation of the xDSL transmission signal above the POTS frequency band 12, and therefore minimizes intermodulation products and noise (resulting from the xDSL transmission) within the POTS band 12. Although only one such shaped signal band 56 is illustrated in FIG. 3D, it will be appreciated that this aspect of the invention is not so limited. Instead, other shapes may be deemed desirable, depending upon the specific environment and line conditions.

Reference is now made to FIG. 4, which shows a block diagram of a modem 20 constructed in accordance with the present invention. As is common among modems, the modem 20 is in communication with both a local loop 26 and computing equipment 25, such as a personal computer. More specifically, the modem 20 communicates with the computing equipment 25 across line 60. The telephone line 26 is typically comprised of a two wire service, which wires are often denoted as TIP 62 and RING 64. The TIP 62 and RING 64 lines are input to an analog front-end circuit 66 (see FIG. 5) as well as a monitor circuit 68, which is configured to detect an OFF-HOOK condition of the local loop 26.

Analog-to-digital and digital-to-analog converter (ADC and DAC, respectively) circuitry 70 is in communication with the analog front end circuitry 66, and is in further communication with digital signal processor 72. Data received from the local loop 26 passes through the analog front-end 66 and is converted from analog-to-digital form by the analog to digital converter of block 70, before being passed to the digital signal processor 72. Conversely, outgoing data output from the digital signal processor 72 is converted by the digital to analog converter of block 70, before being communicated to the local loop 26, by way of the analog front-end 66. Finally, Data Terminal Equipment (DTE) interface 74 is in communication with the digital signal processor 72 and in further communication across line 60, with the data terminal equipment, such as a computer 25. The analog-to-digital and digital-to-analog converter circuitry 70, the digital signal processing 72, and the DTE interface 74 are all well known and generally operate in accordance with the prior art. Therefore, their individual structure and operation need not be described herein.

Indeed, a significant component of the modem 20, constructed in accordance with the present invention, is a controller 80 that is in communication with the various other components of the modem 20. While there are various ways to implement the controller 80, one way, as illustrated, is to further partition the controller 80 into functional units denoted as a processing unit 82, a memory 84 (which may further include an executable code segment 86), and a controller 88.

For purposes of the broad concepts of the present invention, the controller 80 receives a signal from the monitor circuit 68 on line 90, which signal indicates whether the invention should transmit data in a band-limited transmission state or a full-band transmission state. In this regard, the monitor circuitry 68 may be configured to detect an OFF-HOOK condition, or alternatively, a RING condition on local loop 26. As is known in the art, the OFF-HOOK condition may be detected by a drop in voltage across the local loop 26, or alternatively, a sudden chance in impedance on the local loop 26. On the other hand, a RING detect condition is identified by a low frequency oscillatory voltage on local loop 26. For example, the voltage drops from about 48 volts (on hook) to approximately 10 volts or less (off hook), at the customer premises end of the local loop.

In short, the controller 80 evaluates the signal received on line 90 to determine whether data should be transmitted in the full-band transmission state or the band-limited transmission state. Appropriate signals may, accordingly, be transmitted to the digital signal processor 72 for formulating data transmissions (or interpreting received data transmissions).

In accordance with an alternative embodiment of the invention, it will be appreciated that the monitor circuitry 68 may be incorporated within the controller 80, whereby certain signal conditions may be evaluated to detect the band-limiting condition. In this regard, an analog-to-digital converter would also be implemented as part of the controller 80, to generate a signal in digital format which may be more readily evaluated and processed by the processing unit 82. In this regard, the processing unit 82 may be a microprocessor, a microcontroller, an application specific integrated circuit (ASIC), or other digital circuitry configured to specifically process information. In the illustrated embodiment, the controller 80 includes fundamental components (processor unit, controller, memory) that together operate to perform distinct computing operations. Such operations may be controlled, for example, by executable code 86 contained within the memory 84.

Reference is now made to FIG. 5, which shows a more detailed diagram of the circuitry comprising the analog front end element 66. The preferred embodiment includes blocking capacitors 102 and 104, which are series connected with the TIP 62 and RING 64 signal lines, and serve to block any DC voltage otherwise carried on the TIP 62 and RING 64 lines. A transformer 106 couples alternating current to the remainder of the circuitry, as well as provides safety and signal isolation for the remaining circuitry in the modem. A termination resistor 108 and switch 110 are disposed for series connection with each other (depending upon whether the switch 110 is opened or closed), and together are connected in parallel across the secondary winding of the transformer 106. The switch 110 is controlled by controller 80 (FIG. 4) to close and therefore switch in the terminating resistor 108 when the telephones 30 and 32 (see FIG. 2) are all ON-HOOK (as observed by the monitor circuit 68). The switch 110 may be open to switch out the terminating resistor 108, upon detection of an incoming RING signal or OFF-HOOK on the local loop 26. Capacitors 102 and 104 are chosen to pass data, block DC, and yield acceptable Ringer Equivalence Number per FCC part 68. The switch 110 is generally opened to switch out the terminating resistor when the monitor circuit 68 determines that the local loop 26 is in the OFF-HOOK state. The reason for this is that, when one or more telephones are taken OFF-HOOK, then the OFF-HOOK telephone will terminate the line, and the terminating resistor 108 is not needed. Optionally, the switch 110 can be closed in the Off-HOOK state to improve line termination provided by the OFF-HOOK telephone.

The item represented by reference numeral **112** denotes circuitry that is configured in a form of a dependent current source. The current source is prompted by the transmit signal Tx to create an outgoing transmission signal. As a current source, the item **112** has a very high impedance (as seen across the secondary winding of transformer **106**), and therefore, only the termination resistor **108** operates to terminate the line (when switched in). Similarly, amplifier **114** is the receive amplifier that generates the Rx signal, as is known in the art. Like the current source **112**, the amplifier **114** has an extremely high input impedance and thus does not effect line termination.

Reference is now made to FIG. **6**, which shows a software flow-chart illustrating the operation of the analog front-end element **66** of FIG. **5**. Beginning at step **120**, the element **66** determines whether the local loop **26** is ON-HOOK or OFF-HOOK. As will be appreciated from the foregoing discussion, this decision is made by the controller **80**, which outputs a signal **122** (see FIG. **4**) to the analog front-end **66** indicative of the ON-HOOK/OFF-HOOK status. If the resolution of step **120** is NO, the analog front-end element **66** opens switch **112** (step **122**) to remove the termination resistor **108** from the circuit. That is, if the system detects that a telephone connected to the local loop **26** is OFF-HOOK, it will remove the termination resistor **108** from the circuit, since the line will then be terminated by the OFF-HOOK telephone. Thereafter, operation proceeds to step **122**, wherein data is transmitted in accordance with the band-limited transmission frequency band (e.g., 20 kHz–100 kHz). In accordance with one embodiment of the present invention, the system may emit periodic tones within the audible frequency range to alert a user talking on an attached telephone that the local loop **26** is also being used for data transmissions. Thus, a person, for example, speaking in another part of the house over a telephone hearing periodic beeps would know that someone else in the household is using a computer to communicate data, and therefore, may wish to keep his or her conversation to a minimum, in order to free up the local loop **26**, so that the present invention may obtain a full utilization of the full-banded transmission frequency band, for maximum data throughput.

If the resolution of step **120** is YES, indicating that all telephones attached to the local loop **26** are ON-HOOK, then the system ensures that switch **110** is closed thereby placing termination resistor **108** in the circuit, so as to achieve proper line termination (step **130**). Thereafter, the system may transmit data across the local loop utilizing the entire, full-band transmission frequency (i.e., DC to approximately 100 KHz).

Reference is now made to FIG. **7**, which is a software flow-chart illustrating the top-level operation of a system communicating in accordance with the present invention. Beginning at block **140**, the system awaits the initiation of data transmission. This initiation may occur either upon the instruction of a user at the computer **25** (see FIG. **2**), or alternatively, from a remote user that is dialing the phone number of computer **25** to connect up to that computer (this assumes that that computer **25** is in auto answer mode). Once the system has been instructed to begin data communications, it first makes a check (at step **144**) to determine whether the loop is in the OFF-HOOK state. If so, it begins the data communications in the band-limited frequency transition state (step **146**) (e.g., 20 kHz–100 kHz). During the data transmissions, the system will make continuous checks to determine whether the data transmission has ended (step **148**, or whether the band-limiting condition has subsided (step **150**). As previously mentioned, the band-limiting condition is generally identified by the OFF-HOOK detection circuitry. If the end data communications check, at step **148**, resolves to YES, then the system returns to step **140**. If not, the system proceeds to step **150** where it checks for the cessation of the band-limiting condition. If this step resolves to YES, then the system continues the data transmission in the full-band transmission frequency bandwidth (step **154**).

Returning to the decision block **144**, if, upon initiation of data communication, the system determines that all telephones are presently ON-HOOK, then the system proceeds to step **154** where it transmits data in accordance with the full-band data transmission state (i.e., utilizing the full 0 to 100 KHz transmission frequency bandwidth). During transmission in this frequency band, the system periodically checks to see if the data communications has terminated (step **156**), or whether the occurrence of a band-limiting condition has occurred (step **158**). This latter condition occurs, for example, when a person lifts a handset of an attached telephone. If this occurs, the system proceeds to step **146** and continues the data transmissions in accordance with the band-limited transmission frequency band (20 kHz–100 kHz).

It will be appreciated from a review of the flow-chart of FIG. **7**, that the system, during data transmission, can dynamically shift back and forth between the full-band and band-limited transmission frequency bandwidths as users may lift or reset telephone handsets (or as RING conditions occur). It will be appreciated, however, that other band-limiting conditions (other than RING or OFF-HOOK) may be utilized to invoke the frequency shifting feature of the present invention, depending upon the system configuration or other pertinent system factors.

It will be appreciated that the invention described herein could provide a low-cost solution to Internet access for the mass consumer market. In this regard, it could fill the gap between low-cost 33.6 kbps modems and high speed xDSL modems, which require the addition of relatively expensive equipment (such as POTS splitters and POTS filters) at the customer premise, and is labor intensive The present invention, as described above, generally achieves transmission rates in the range of 64 kbps to 640 kbps.

As described above, the invention utilizes the low frequency portion of the telephone subscriber loop spectrum (roughly DC to approximately 100 KHz) to transport user data. The modulation could be CAP (carrierless amplitude-phase), QAM (quadrature amplitude modulation), DMT (dual multi-tone), spread spectrum, etc, as the invention is not limited to any particular form. Utilization of the lower frequency portion of the telephone subscriber loop has the advantage of the lowest possible signal attenuation (usually the number one signal impairment in data communications) and low cross-talk. Other advantages are reduced transmission line concerns like reflections due to stubs.

In use, the invention requires a simple bridge (electrical parallel) connection to the subscriber loop or premise wiring. Therefore, one unit would connect (in bridge fashion) at the central office, and one companion unit connect at the customer premises.

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.

We claim:

1. A modem for communicating across a communication link comprising:

an input/output signal line in communication with the communication link;

a processor unit adapted for operation in one of at least two states, a full-band transmission state and a band-limited state, wherein the full-band transmission state is defined by significant transmission energy in a frequency range below a first frequency, and a band-limited transmission state defined by a negligible amount of energy in the frequency range below the first frequency and a significant amount of energy above the first frequency, wherein the first frequency is at least 4000 Hertz.

2. The modem as defined in claim 1, wherein the first frequency is approximately 15 kilohertz.

3. The modem as defined in claim 1, wherein the communication link is a multiple-use communication link.

4. The modem as defined in claim 1, further including:

sensing means for sensing a band-limiting condition; and

control means associated with the processor unit responsive to the sensing means for controlling the operating state of the processor unit, wherein upon sensing the band-limiting condition the control means causes the processor to operate in the band-limited state, and upon sensing no band-limiting condition the control means causes the processor to operate in the full-band transmission state.

5. The modem as defined in claim 1, wherein significant energy transmissions are transmissions substantially exceeding an audible level.

6. A modem for communicating across a communication link capable of single-use transmissions and multiple-use transmissions comprising:

an input/output signal line in communication with the communication link;

a processor unit adapted for operation in one of at least two states, a full-band transmission state and a band-limited state, the full-band transmission state defined by data transmission in a frequency band including frequencies above and below approximately 4000 Hertz and the band-limited state defined by data transmission in a frequency band including frequencies only above approximately 4000 Hertz, wherein the full-band transmission state occurs when single-use transmissions are occurring across the transmission link, and the band-limited transmission state occurs when multiple-use transmissions are occurring across the communication link.

7. A modem for communicating across a communication link comprising:

an input/output signal line in communication with the communication link;

a processor unit adapted for operation in one of at least two states, a full-band transmission state and a band-limited state, wherein the full-band transmission state is defined by significant energy transmission below a first frequency and the band-limited state is defined by substantially zero energy transmission below the first frequency wherein the first frequency is at least 4000 Hertz.;

sensing means for sensing a band-limiting condition; and

control means associated with the processor unit responsive to the sensing means for controlling the operating state of the processor unit, wherein upon sensing the band-limiting condition the control means causes the processor to operate in the band-limited state, and upon sensing no band-limiting condition the control means causes the processor to operate in the full-band transmission state.

8. The modem as defined in claim 7, wherein the first frequency is approximately 15 kilohertz.

9. The modem as defined in claim 7, wherein the sensing means is configured to detect a multi-position switch, the position of which defines the band-limiting condition.

10. The modem as defined in claim 7, wherein the sensing means is configured to detect an off-hook condition of a telephone that is electrically connected to the input/output signal line.

11. The modem as defined in claim 10, wherein the sensing means further includes means for detecting the onset of a condition indicative of a handset of the telephone being taken off-hook.

12. The modem as defined in claim 11, wherein the means for detecting the onset of the condition is configured to detect a voltage drop on the input/output signal line.

13. The modem as defined in claim 11, wherein the means for detecting the onset of the condition is configured to detect an impedance shift in the input/output signal line.

14. The modem as defined in claim 7, wherein the full-band transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of less than 4 kilohertz.

15. The modem as defined in claim 14, wherein the full-band transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of approximately DC.

16. The modem as defined in claim 7, wherein the full-band transmission state is defined by a transmission frequency bandwidth having an upper frequency boundary of greater than 50 kilohertz.

17. The modem as defined in claim 16, wherein the full-band transmission state is defined by a transmission frequency bandwidth having an upper frequency boundary of approximately 100 kilohertz.

18. The modem as defined in claim 7, wherein the band-limited transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of greater than 4 kilohertz.

19. The modem as defined in claim 18, wherein the full-band transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of approximately 20 kilohertz.

20. The modem as defined in claim 7, wherein the full-band transmission state is defined by a first transmission frequency bandwidth and the band-limited transmission state is defined by a second transmission frequency bandwidth, wherein the first transmission frequency bandwidth has an upper frequency boundary that is substantially the same an upper frequency boundary of the second frequency bandwidth.

21. The modem as defined in claim 7, wherein the full-band transmission state is defined by a first transmission frequency bandwidth and the band-limited transmission state is defined by a second transmission frequency

bandwidth, wherein the first transmission frequency bandwidth has an upper frequency boundary that is different than an upper frequency boundary of the second frequency bandwidth.

22. The modem as defined in claim 7, wherein the communication link is a two-wire telecommunications link.

23. The modem as defined in claim 7, wherein the communication link is a local loop.

24. The modem as defined in claim 7, wherein the sensing means includes a code segment containing executable code.

25. The modem as defined in claim 7, wherein the control means includes a code segment containing executable code.

26. A modem for communicating across a communication link comprising:

an input/output signal line in communication with the communication link;

a processor unit adapted for operation in one of two states, a full-band transmission state and a band-limited state, wherein the full-band transmission state is defined by a lower frequency boundary below a second frequency and an upper frequency boundary greater than a first frequency, and the band-limited state is defined by a lower frequency boundary greater than the second frequency and an upper frequency boundary greater than the first frequency, wherein the first frequency is approximately 4000 Hertz;

a sensor configured to detect the presence of a band-limiting condition; and

a controller associated with the processor unit and responsive to the sensor, configured to control the operating state of the processor unit, wherein upon sensing the band-limiting condition the controller causes the processor to operate in the band-limited state, and upon sensing no band-limiting condition the controller causes the processor to operate in the full-band transmission state.

27. The modem as defined in claim 26, wherein the first frequency is approximately 50 kilohertz.

28. The modem as defined in claim 26, wherein the second frequency is approximately 4 kilohertz.

29. A method for dynamically communicating data over a communication link using a modem comprising the steps of transmitting data in a full-band transmission state, wherein the full-band transmission state is defined by a frequency band having a lower boundary below 4000 Hertz and an upper boundary above 4000 Hertz;

sensing a band-limiting condition; and

adjusting the transmission of data from the full-band transmission state to a band-limited transmission state, in response to the sensing step, wherein the band-limited transmission state is defined by a frequency band entirely above 4000 Hertz.

30. The method as defined in claim 29, wherein the sensing step includes detecting the position of a multi-position switch.

31. The method as defined in claim 29, further including the step of adaptively varying transmit power of the transmission of data to minimize interference of data signals with a lower frequency band.

32. The method as defined in claim 29, further including the step of uniquely shaping a power spectral transmission band of the data transmission to minimize interference of data signals with a lower frequency band.

33. The method as defined in claim 29, further including the step of sensing a cessation of the band-limiting condition.

34. The method as defined in claim 33, further including the step of adjusting the transmission of data from the band-limited transmission state to the full-band transmission state, in response to the step of sensing the cessation of the band-limiting condition.

35. The method as defined in claim 29, wherein the step of sensing the band-limiting condition includes sensing an incoming ring signal on the communication link.

36. The method as defined in claim 29, wherein the step of sensing a band-limiting condition includes sensing an off-hook condition of a telephone handset of a telephone electrically connected to the communication link.

37. The method as defined in claim 36, wherein the step of sensing the off-hook condition includes sensing an impedance of the communication link.

38. The method as defined in claim 36, wherein the step of sensing the off-hook condition includes sensing a voltage on the communication link.

39. The method as defined in claim 29, wherein the full-band transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of less than 4 kilohertz.

40. The method as defined in claim 39, wherein the full-band transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of approximately DC.

41. The method as defined in claim 29, wherein the full-band transmission state is defined by a transmission frequency bandwidth having an upper frequency boundary of greater than 50 kilohertz.

42. The method as defined in claim 41, wherein the full-band transmission state is defined by a transmission frequency bandwidth having an upper frequency boundary of approximately 100 kilohertz.

43. The method as defined in claim 29, wherein the band-limited transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of greater than 4 kilohertz.

44. The method as defined in claim 43, wherein the full-band transmission state is defined by a transmission frequency bandwidth having a lower frequency boundary of approximately 20 kilohertz.

45. The method as defined in claim 29, wherein the full-band transmission state is defined by a first transmission frequency bandwidth and the band-limited transmission state is defined by a second transmission frequency bandwidth, wherein the first transmission frequency bandwidth has an upper frequency boundary that is substantially the same an upper frequency boundary of the second frequency bandwidth.

46. The method as defined in claim 29, wherein the full-band transmission state is defined by a first transmission frequency bandwidth and the band-limited transmission state is defined by a second transmission frequency bandwidth, wherein the first transmission frequency bandwidth has an upper frequency boundary that is different than an upper frequency boundary of the second frequency bandwidth.

47. A method for dynamically communicating data across a communication link using a modem comprising the steps of:

transmitting data in a band-limited transmission state, wherein the band-limited transmission state is defined by a frequency band entirely above 4000 Hertz;

sensing a cessation in a band-limiting condition; and

adjusting the transmission of data from the band-limited transmission state to a full-band transmission state, in response to the sensing step, wherein the full-band transmission state is defined by a frequency band extending at least partially below 4000 Hertz.

48. The method as defined in claim **47**, further including the step of sensing a band-limiting condition.

49. The method as defined in claim **48**, further including the step of adjusting the transmission of data from the full-band transmission state to the band-limited transmission state, in response to the step of sensing the band-limiting condition.

50. A modem for communicating across at least two channels of a communication link comprising:

means for defining a first communication channel having a dynamic bandwidth, the first communication channel having a full-band transmission state and a band-limited transmission state, wherein the full-band transmission state is defined by a frequency band that at least partially extends below 4000 Hertz, and the band-limited transmission state is defined by a frequency band that is entirely above 4000 Hertz;

means for accommodating a second communication channel;

means for controlling the communication across the first communication channel such that the communication is in the full-band transmission state when no communications are occurring across the second communication channel, and the communication is in the band-limited transmission state when communications are occurring across the second communication channel.

51. The modem as defined in claim **50**, wherein the second communication channel is a POTS communication channel.

52. The modem as defined in claim **51**, wherein information transmitted across the second communication channel is voice information.

53. The modem as defined in claim **51**, wherein information transmitted across the second communication channel is data communicated from a PSTN modem.

54. A modem for communicating across at least two channels of a communication link comprising:

circuitry configured to define a first communication channel having a dynamic bandwidth, the first communication channel having a full-band transmission state defined by a frequency band that extends at least partially below 4000 Hertz, and a band-limited transmission state, defined by a frequency band that exists entirely above 4000 Hertz;

circuitry configured to accommodate a second communication channel;

circuitry configured to control the communication across the first communication channel such that the communication is in the full-band transmission state when no communications are occurring across the second communication channel, and the communication is in the band-limited transmission state when communications are occurring across the second communication channel.

* * * * *

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 6,061,392
DATED : May 9, 2000
INVENTOR(S) : Bingel et al.

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 1,
Line 24, after the word "modems" add the symbol -- , --.

Column 2,
Line 3, after the phrase "Accordingly," delete the word "their" and substitute therefor -- there --.

Column 3,
Line 30, after the word "e.g." add the symbol -- , --.
Line 36, after the word "bands" delete the word "to" and substitute therefor -- so --.

Column 4,
Line 14, after the word "Another" delete the symbol ",".
Line 39, after the word "without" delete the word "out".

Column 6,
Lines 8 and 9, after the word "the" delete the phrase "operation of the".
Line 31, after the word "defined" delete the symbol ",".
Line 40, after the word "as" delete the phrase "'xDSL'" and substitute therefor -- "xDSL," --.
Line 47, after the word "than" delete the letter "I" and substitute therefor -- 1 --.

Column 7,
Line 21, after the word "keeping" delete the word "within" and substitute therefor -- with --.
Line 30, after the word "computer" add the numeral -- 25 --.
Line 52, after the second occurrence of the word "the" delete the word "mode" and substitute therefor -- modem --.

Column 8,
Line 43, after the word "and" delete the phrase "thus realize" and substitute therefor -- thus, realizes --.

Column 9,
Line 14, after the word "taper" delete the symbol ",".

Column 10,
Line 8, after the word "sudden" delete the word "chance" and substitute therefor -- change --.
Line 26, after the word "converter" add the numeral -- 70 --.

UNITED STATES PATENT AND TRADEMARK OFFICE
CERTIFICATE OF CORRECTION

PATENT NO. : 6,061,392
DATED : May 9, 2000
INVENTOR(S) : Bingel et al.

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 10, cont'd,
Lines 33 and 34, after the "components" delete the phrase "(processor unit, controller, memory)" and substitute therefor -- (processor unit 82, controller, 88, memory 84) --.
Line 66, afte the word "the" delete the word "Off-HOOK" and substitute therefor -- OFF-HOOK --.

Column 11,
Line 9, after the second occurrence of the word "the" delete the phrase "Rx signal" and substitute therefor -- receive signal, Rx --.
Line 12, after the word "not" delete the word "effect" and substitute therefor -- affect --.
Line 19, after the word "signal" delete the numeral "122" and substitute therefor -- 121 --.
Line 22, after the word "switch" delete the numeral "112" and substitute therefor -- 110 --.
Line 33, after the word "telephone" add the phrase -- 30, 32 (see FIG. 2) --.
Line 37, after the word "computer" add the phrase -- 25 (see FIG. 2) --.
Line 43, after the word "telephones" add the phrase -- 30, 32 (see FIG. 2) --.
Line 66, after the numeral "148" add thesymbol --) --.

Column 12,
Line 20, after the word "telephone" add the phrase -- 30, 32 (see FIG. 2) --.
Line 28, after the word "telephone" add the phrase -- 30, 32 (see FIG. 2) --.
Lines 48 and 49, after the word "DMT" delete the word "dual" and substitute therefor -- discrete --.
Line 49, after the word "etc" add the symbol -- . --.

Column 13,
Line 12, after the phrase "link;" add the word -- and --.
Line 34, after the word "processor" add the word -- unit --.
Line 36, after the word "processor" add the word -- unit --.
Line 55, after the first occurrence of the word "the" delete the word "transmission" and substitute therefore -- communication --.
Line 56, after the word "band-limited" delete the word "transmission".

Column 14,
Line 2, after the word "frequency" add the symbol --, --.
Line 9, after the word "processor" add the word -- unit --.
Line 11, after the word "processor" add the word -- unit --.
Line 37, after the word "by" delete the word "a" and substitute therefor -- the --.
Line 38, after the word "having" delete the word "a" and substitute therefor -- the --.

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 6,061,392
DATED : May 9, 2000
INVENTOR(S) : Bingel et al.

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 14, cont'd,
Line 45, after the word "by" delete the word "a" and substitute therefor -- the --.
Line 46, after the word "having" delete the word "an" and substitute therefor -- the --.
Line 53, after the word "by" delete the word "a" and substitute therefor -- the --.
Line 54, after the word "having" delete the word "a" and substitute therefor -- the --.
Line 58, after the word "band-limited" delete the word "transmission".
Line 62, after the word "same" add the word -- as --.
Line 66, after the word "band-limited" delete the word "transmission".

Column 15,
Lines 33, 34 and 36, after the word "processor" add the word -- unit --.
Line 43, after the word "of" add the symbol -- : --
Line 44, indent the word "transmission".

Column 16,
Line 24, after the word "by" delete the word "a" and substitute therefor -- the --.
Line 25, after the word "having" delete the word "a" and substitute therefor -- the --.
Line 32, after the word "by" delete the word "a" and substitute therefor -- the --.
Line 33, after the word "having" delete the word "an" and substitute therefor -- the --.
Line 40, after the word "by" delete the word "a" and substitute therefor -- the --.
Line 41, after the word "having" delete the word "a" and substitute therefor -- the --.
Line 49, after the word "same" add the word -- as --.
Line 49, after the word "second" add the word -- transmission --.

Column 17,
Line 5, after the word "sensing" delete the word "a" and substitute therefor -- the --.
Line 25, after the word "channel" add the word -- and --.

Column 18,
Line 21, after the word "channel" add the word -- and --.

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 6,061,392
DATED : May 9, 2000
INVENTOR(S) : Bingel et al.

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Replace FIG. 6 with:

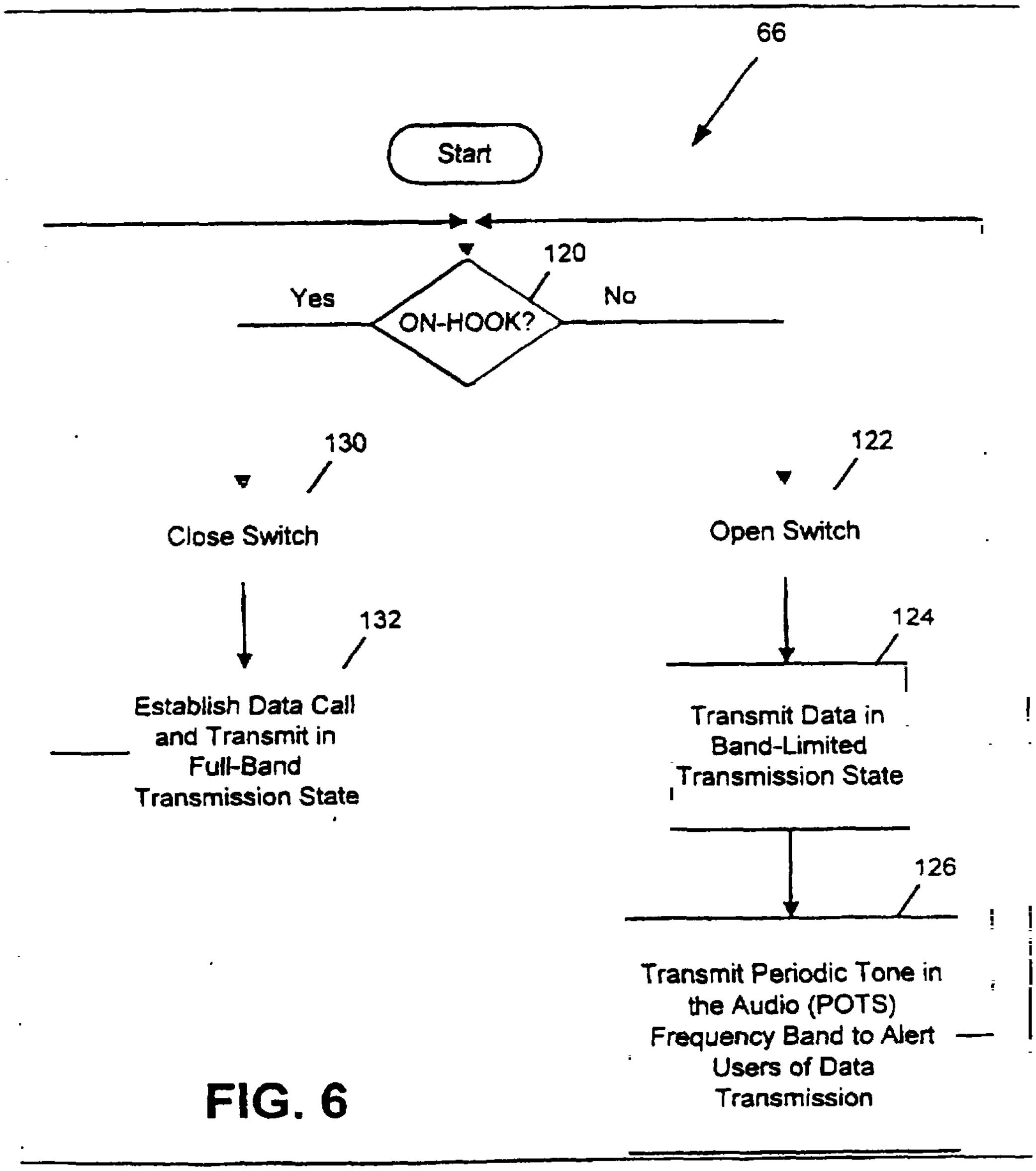

FIG. 6

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 6,061,392
DATED : May 9, 2000
INVENTOR(S) : Bingel et al.

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Replace FIG 4 with:

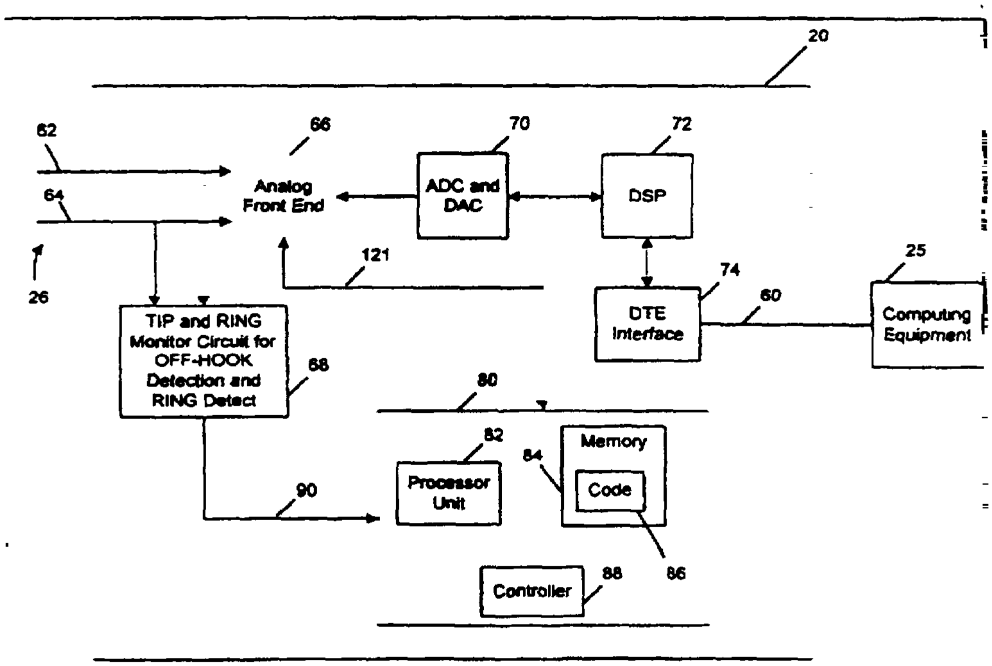

FIG. 4

Signed and Sealed this

Fifth Day of February, 2002

Attest:

JAMES E. ROGAN
Director of the United States Patent and Trademark Office

Attesting Officer

EXHIBIT G
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US005826034A

United States Patent [19]

Albal

[11] **Patent Number: 5,826,034**

[45] **Date of Patent: Oct. 20, 1998**

[54] **SYSTEM AND METHOD FOR TRANSMISSION OF COMMUNICATION SIGNALS THROUGH DIFFERENT MEDIA**

[75] Inventor. **Nandakishore A. Albal**, Largo, Fla.

[73] Assignee: **Paradyne Croporation**, Largo, Fla.

[21] Appl. No.. **695,033**

[22] Filed: **Aug. 9, 1996**

[51] **Int. Cl.**[6] **G06F 5/00**

[52] **U.S. Cl.** **395/200.69**

[58] **Field of Search** .... 364/DIG. 1, DIG. 2; 395/200.69, 761, 500, 326, 329, 180, 200.3, 200.5, 200.57, 200.62, 200.68; 340/825.03, 268.01, 284.3

[56] **References Cited**

U.S. PATENT DOCUMENTS

4,837,798 6/1989 Cohen et al. 379/88

5,381,527 1/1995 Inniss et al. 395/200 69

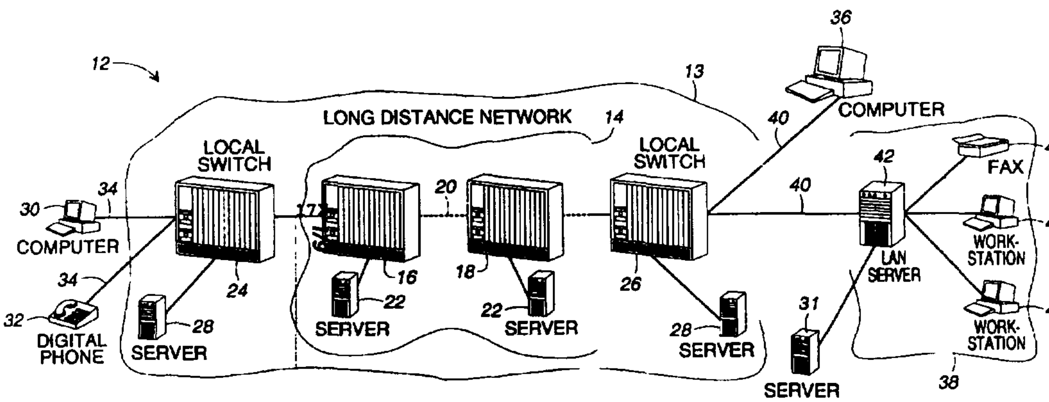
Primary Examiner—Robert B. Harrell

Attorney, Agent, or Firm—Thomas, Kayden, Horstemeyer & Risley, L.L.P.

[57] **ABSTRACT**

An end-to-end ubiquitous payload delivery system and method transfers a payload using multiple communication method following rules established by a sender. The payload transfer is attempted using a preferred media until the parameters of the transfer are exceeded (i.e., a certain number of attempts in a given duration of time), after which one or more alternative media are used until the payload transfer is completed. Prior to the completion of the transfer, the sender has the capability of querying the payload delivery system to determine the status of the payload. Upon completion of the delivery, the sender receives notification.

37 Claims, 10 Drawing Sheets



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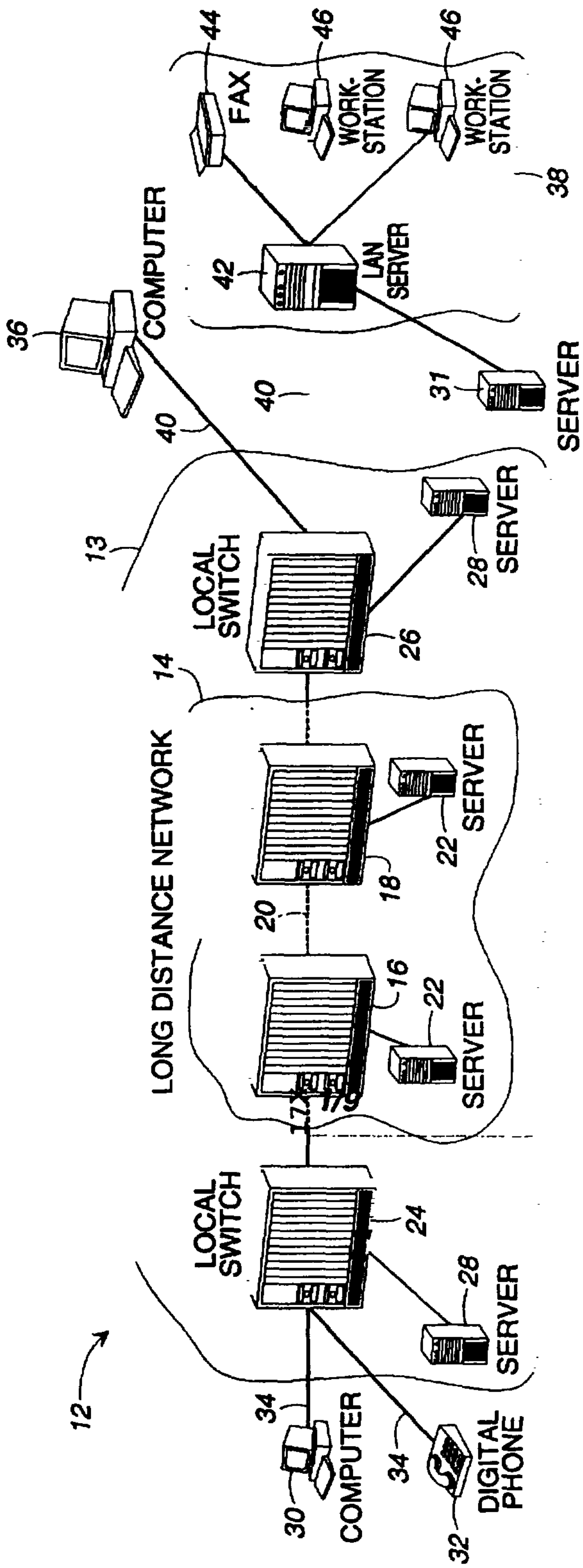

FIG. 1

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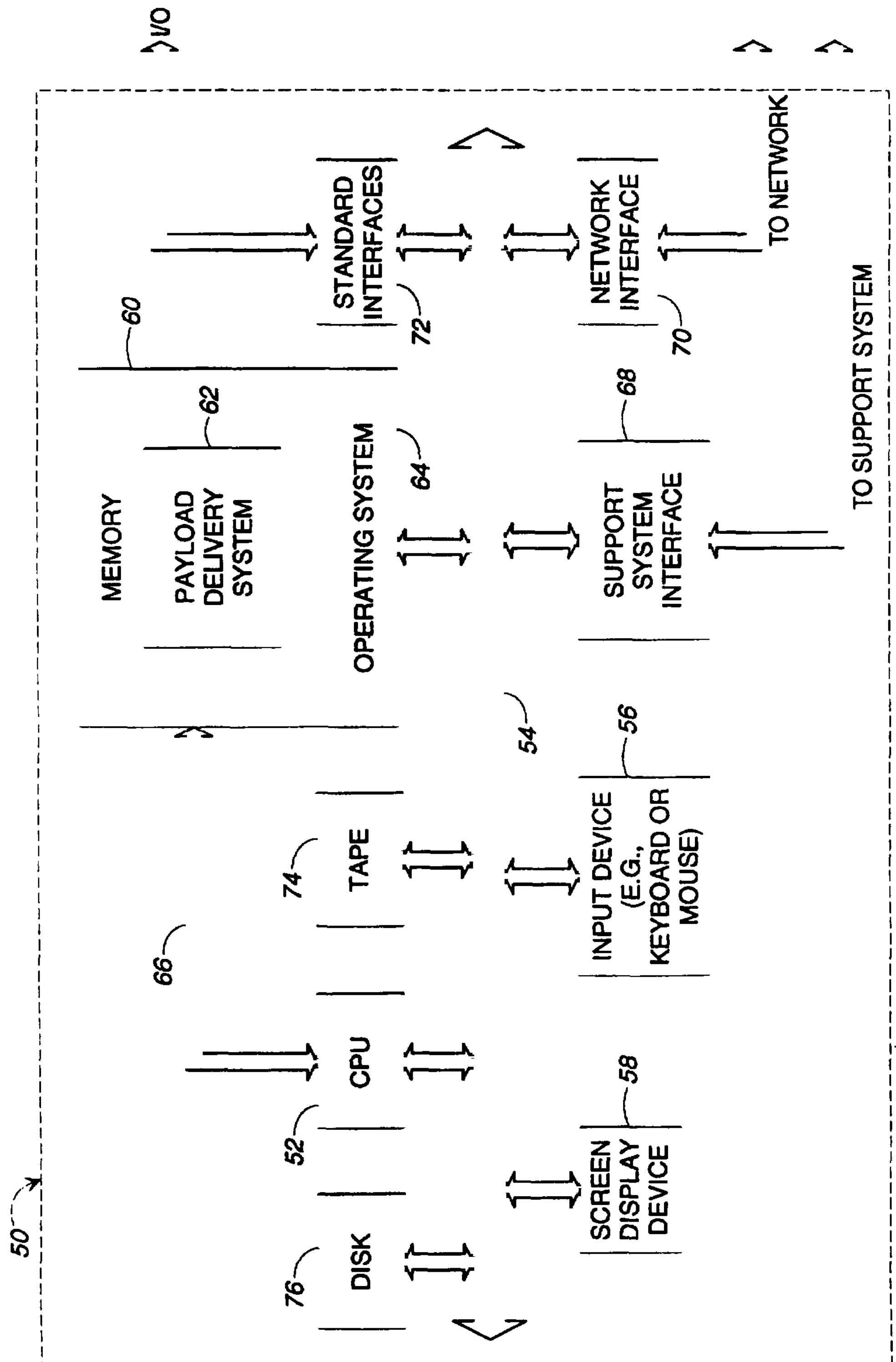

FIG. 2

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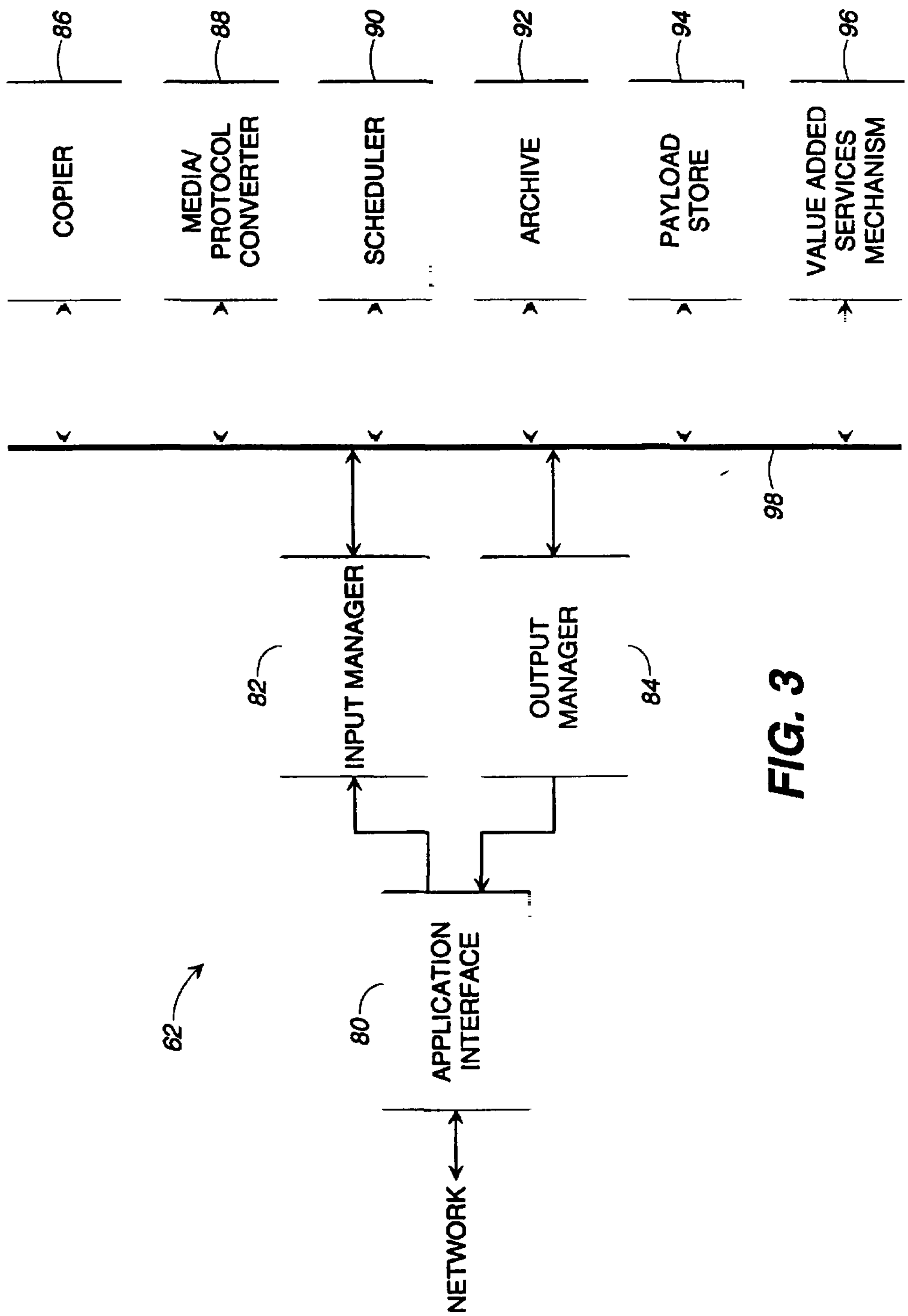

FIG. 3

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NAME: John Doe	
E-MAIL: John@internet.com	Number of Retries: before Media Change 10
	Interval between Retries: : 5 min
	Secondary Media: Fax
	Tertiary Media: Fax-Print
FAX: (770)555-0000	Number of Retries: before Media Change 10
FAX-Print: (770)555-1111	Interval between Retries: : 15 min
	Secondary Media: Fax-Print
	Tertiary Media: None
Address: 120 Sunny Lane	Interval between Retries: : 15 min
Atlanta, Georgia 10002	Secondary Media: None

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FIG. 4

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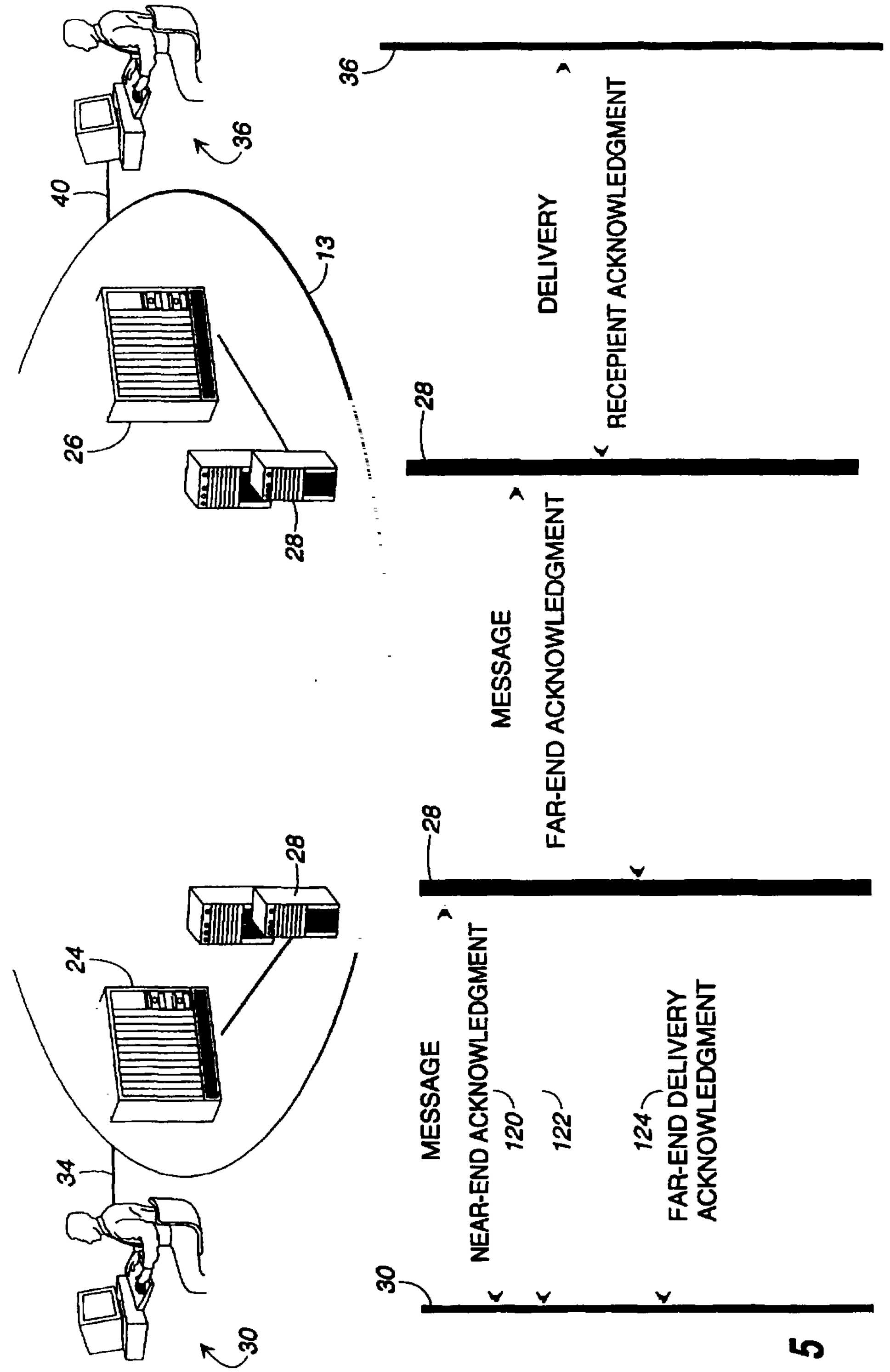

FIG. 5

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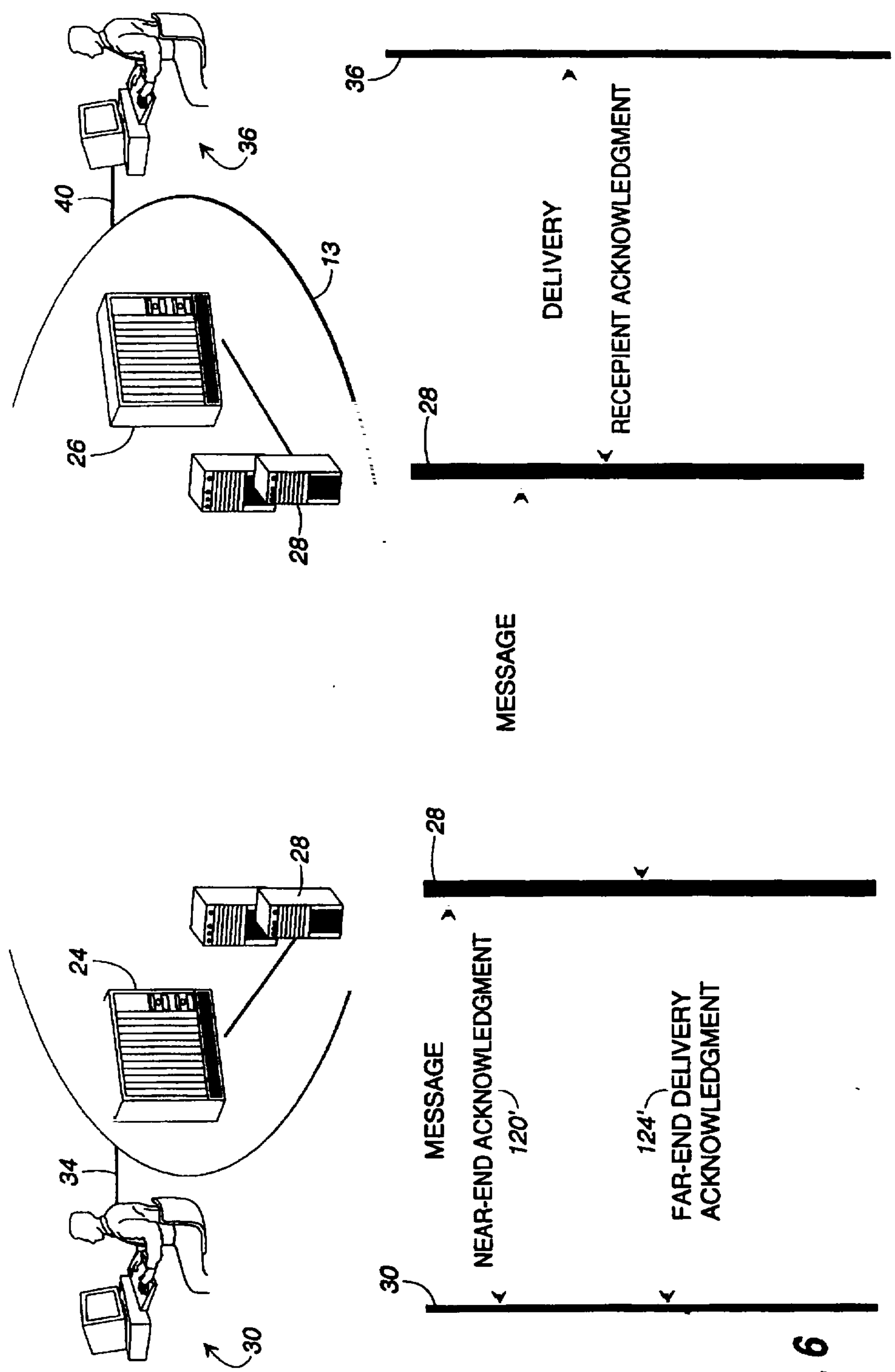

FIG. 6

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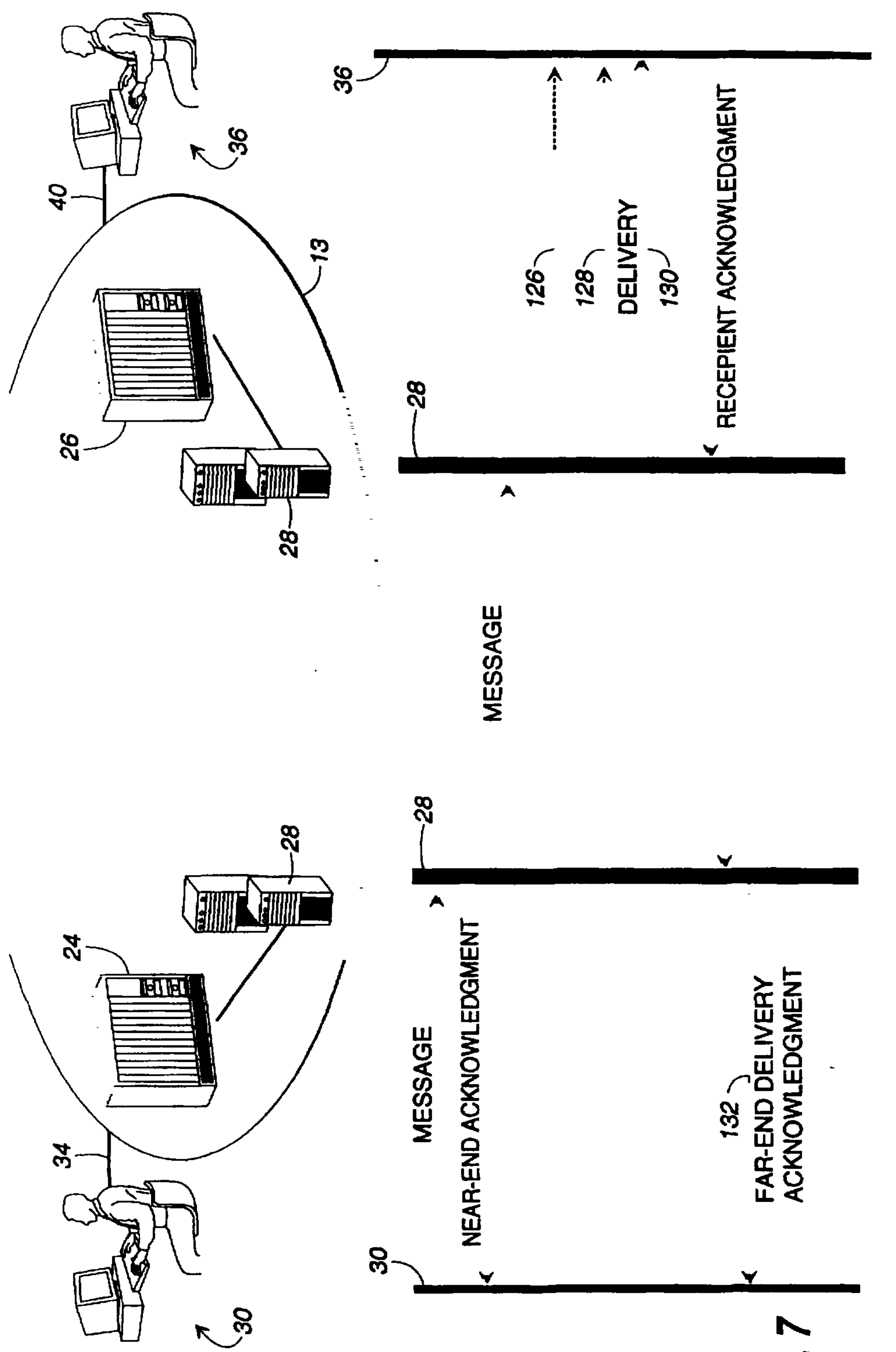
36
40
13
26
28
126
128
DELIVERY
130
RECEPIENT ACKNOWLEDGMENT
MESSAGE
24
34
30
MESSAGE
NEAR-END ACKNOWLEDGMENT
132
FAR-END DELIVERY
ACKNOWLEDGMENT

FIG. 7

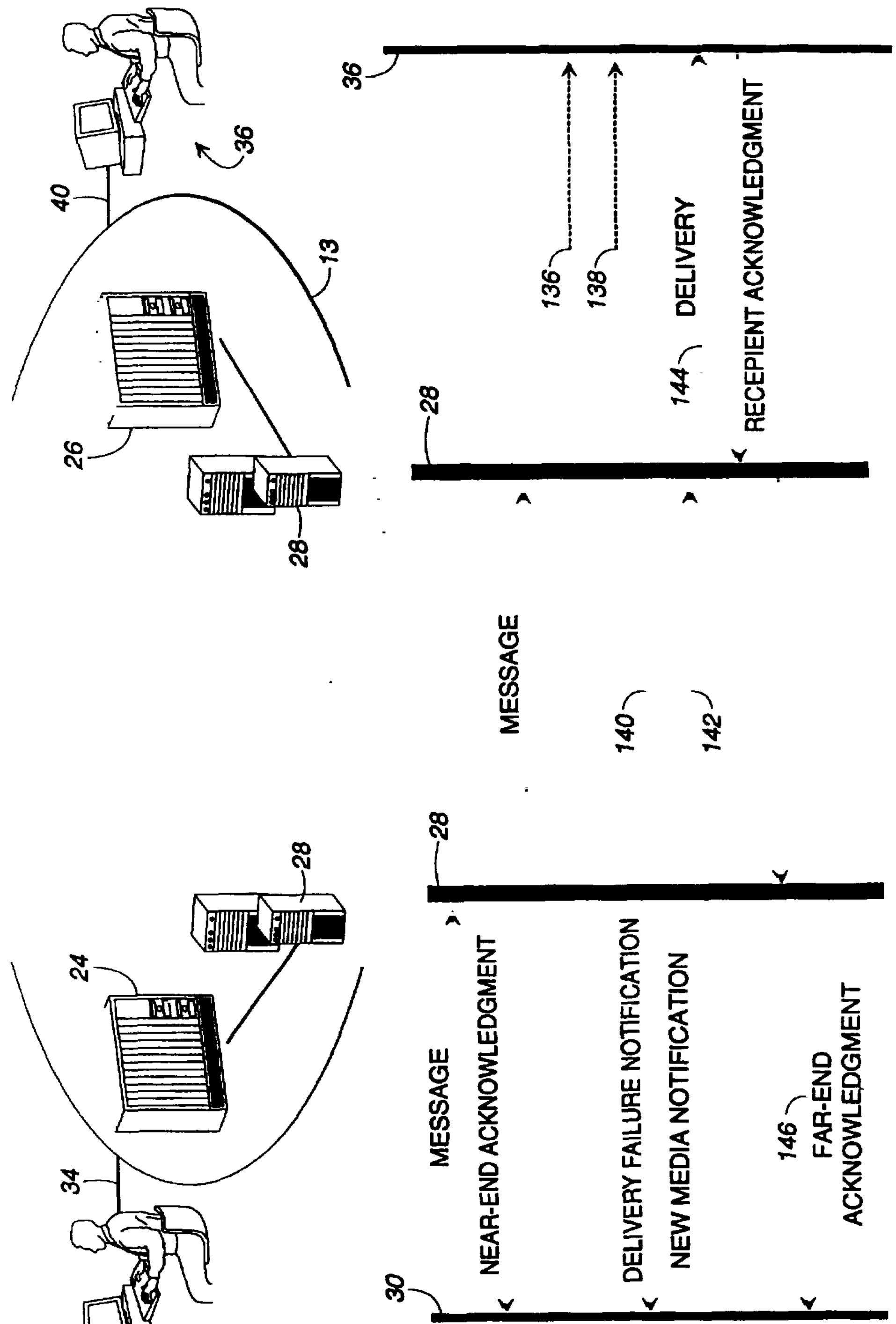

FIG. 8

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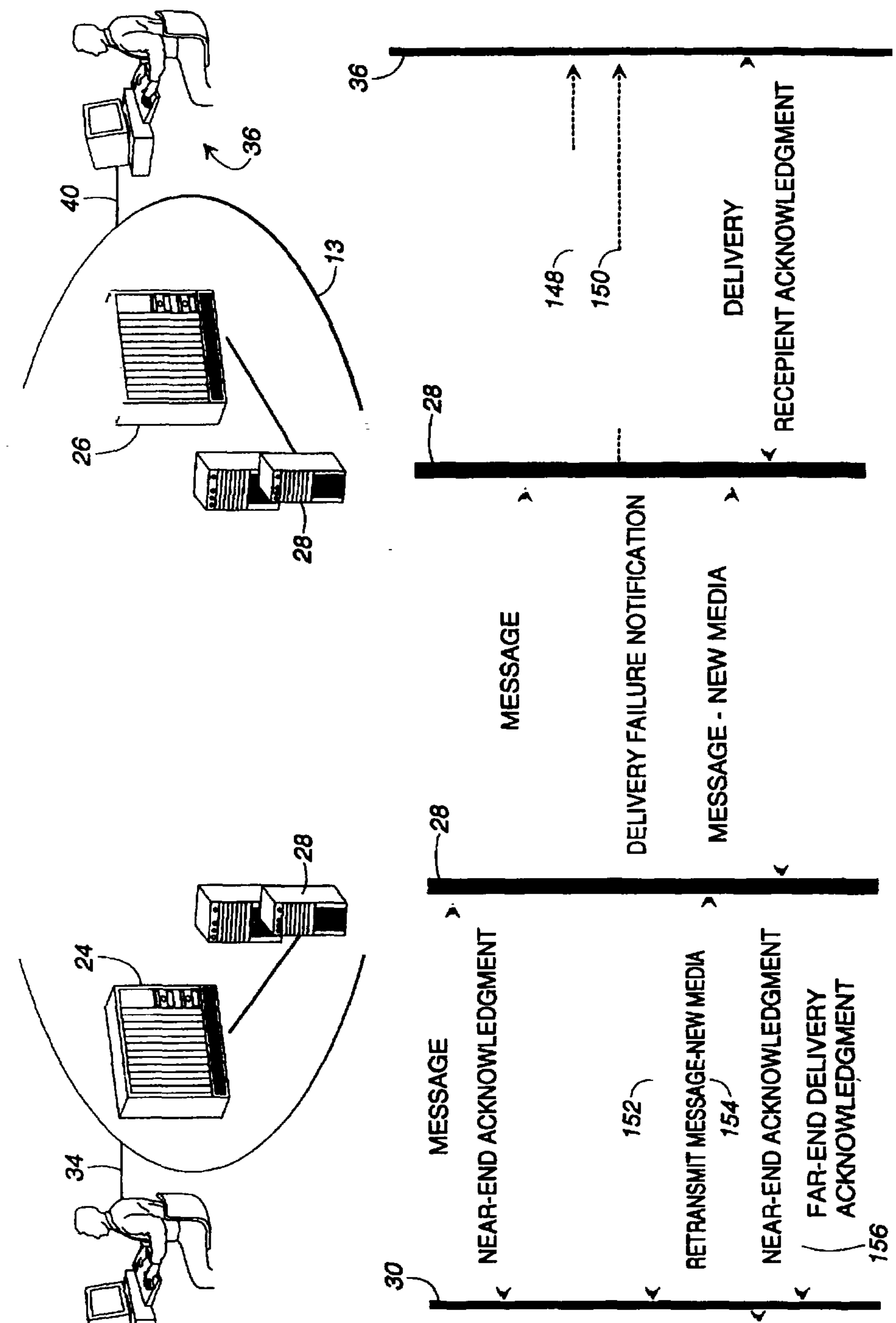
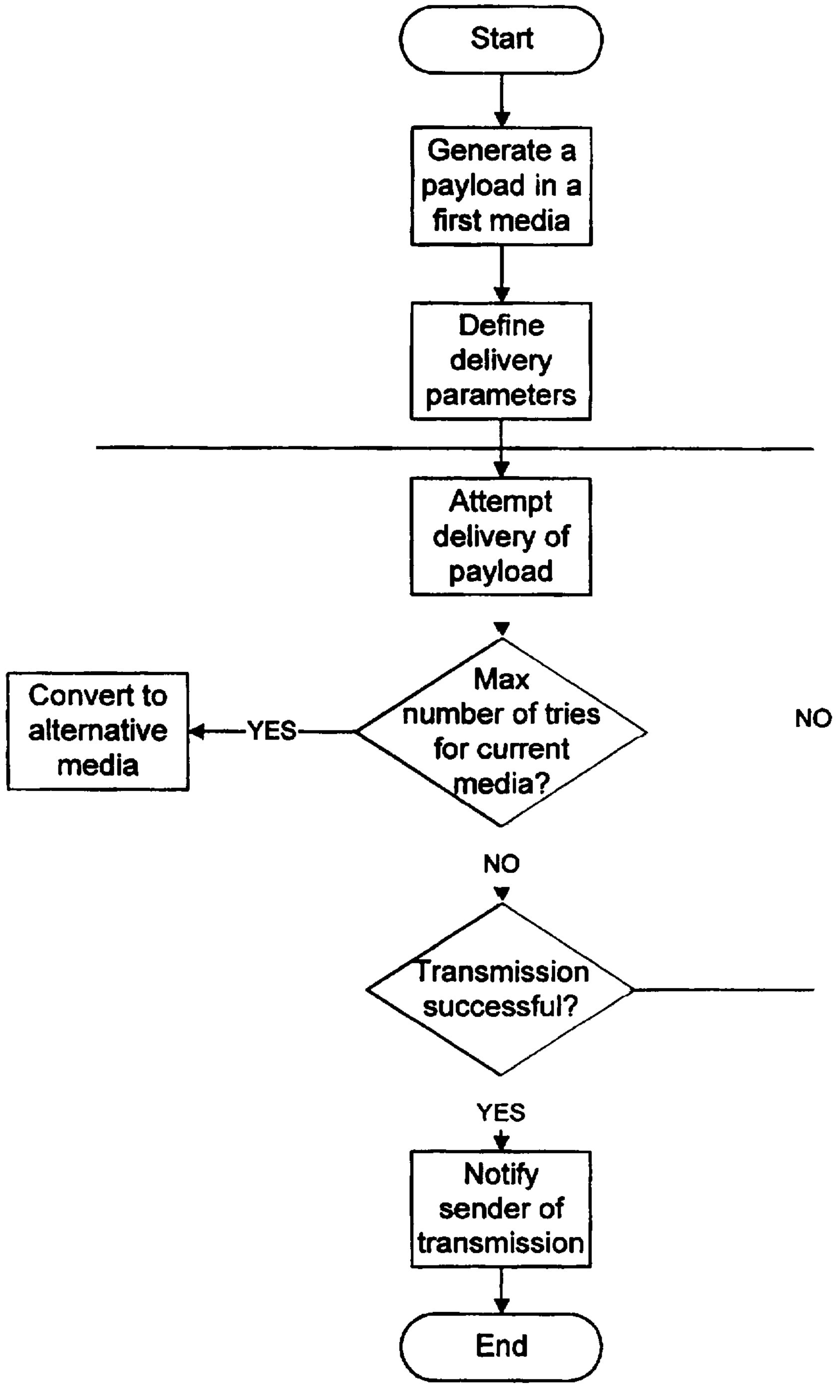


36
40
13
26
28
24
34
30
148
150
MESSAGE
DELIVERY FAILURE NOTIFICATION
MESSAGE - NEW MEDIA
DELIVERY
RECEPIENT ACKNOWLEDGMENT
NEAR-END ACKNOWLEDGMENT
152
RETRANSMIT MESSAGE-NEW MEDIA
154
FAR-END DELIVERY ACKNOWLEDGMENT
156

FIG. 9



Start
Generate a payload in a first media
Define delivery parameters
Attempt delivery of payload
Max number of tries for current media?
YES
Convert to alternative media
NO
NO
Transmission successful?
YES
Notify sender of transmission
End

FIG. 10

SYSTEM AND METHOD FOR TRANSMISSION OF COMMUNICATION SIGNALS THROUGH DIFFERENT MEDIA

FIELD OF THE INVENTION

The present invention generally relates to payload delivery and, more particularly, to an end-to-end payload delivery system and method that effectuates delivery in a media independent manner following the parameters of the delivery that are defined by the sender.

BACKGROUND OF THE INVENTION

Over the ages, the types of media by which people have shared information have changed in stride with advancements in technology, as is especially evident in the present electronic age. Consequently, today there are numerous media for transferring information in a faster and in a more efficient manner than ever before. Examples of such media presently being used include telephone (voice-mail), e-mail, fax, etc., each of which has its own advantages and disadvantages. As a result, in the current competitive market place, reliable communication and the choice of media has come to play a critical role in the success (if not survival) of many businesses, especially those that are geographically diversified. Particularly, the ability to communicate specific information to a person or entity in a reliable, cost effective, and efficient manner is now more of a necessity than a luxury Moreover, all indications are that this ability to communicate will only increase in importance in the coming years as an individual's time becomes more costly because businesses are driven to even greater efficiencies, and as the Internet and the Information Superhighway (e.g., the National Information Infrastructure (NII) or the Global Information Infrastructure (GII)) become globally accessible.

A problem created by having all these different media of communication available is the inability to communicate between the different media. Presently, several communication systems exist that allow a recipient to receive communications in a limited number of different media and then to convert them into a native media. An example of such a system is disclosed in U.S. Pat. No. 4,837,798, issued on Jun. 6, 1989, to Cohen et al., which provides for a single electronic mailbox for receiving messages in different media such as telephone or fax. In the patent to Cohen et al, the unified message system located at the recipient's end converts all the received messages in the user's electronic mailbox into a single native media. This system provides a certain amount of versatility in that the user can receive messages in a given media and covert those messages into a native media of the user. However, the media conversion only occurs post-delivery which prevents the sender from taking advantage of tariffs and competitive service offerings across available media. Further, this system fails to provide any one of the following: acknowledgment or notification to the sender that the recipient actually received the message, acknowledgment or notification to the sender of the success or failure of the message conversion, or a retry mechanism.

Therefore, a heretofore unaddressed need exists in the industry for a payload delivery system that eliminates the incompatibility between different communication services employing different media for communicating information, and that enables the sender to designate the delivery parameters as well as provides notification to the sender when the recipient receives the payload, notification to the sender if

the delivery was not successful, including a request for additional instructions in such a situation, and that allows the sender to query the status of the messages sent.

SUMMARY OF THE INVENTION

The present invention overcomes the inadequacies and deficiencies of the prior art as discussed herein before and as well known in the industry. The present invention provides for a system and method for end-to-end ubiquitous payload delivery that is essentially the electronic equivalent to registered mail with the advantages of speed, configurability, convenience, resource conservation, timeliness, but without the drawbacks of the manual system used with registered mail, e.g., paperwork, delay, time utilization, and geographic limitation of applicability. By utilizing the payload delivery system of the present invention, a sender is able to establish delivery parameters that are complimented by a smart delivery system so as to ensure that the payload is delivered within the set parameters. The delivery parameters preferably include a preferred media of delivery and a number of attempts within a given period of time before conversion of the payload to an alternate media that also has a number of attempts in a given period of time designated before yet further media conversions are performed If required, media conversions and payload copying can be performed at one or more locations where the system resides in order to take advantage of tariffs, special offerings, etc., and to provide guaranteed delivery in a media independent environment. Further, the sender can designate events that trigger notification during delivery of the payload so that the sender is able to keep track of the delivery and receipt of the payload. Accordingly, the sender who knows the makeup of the payload and who traditionally bears the cost of delivery can tailor the payload delivery in order to guarantee receipt and to ensure that the delivery is effectuated in a cost effective and efficient manner.

Briefly stated, an end-to-end ubiquitous payload delivery system in accordance with the present invention comprises a computer program that can be located at any one or more of the following: a senders desktop workstation, a server at the sender's end, a server at the receiver's end, a server connected to a digital switch at either the sender's or recipient's end, or a server in the Internet environment. As a computer application, the end-to-end ubiquitous payload delivery system allows different communication service applications on different media (e.g., e-mail, voice mail, or fax over twisted-pair, coax, untethered/wireless, fiber media) to interoperate across network lines regardless of the underlying communication protocols, operating systems, or databases. This is achieved by converting the entire payload, or a portion of the payload, from its original media to one or more other media as required to complete delivery of the payload in accordance with the delivery parameters defined by the sender. The conversion is performed by the payload delivery system, and therefore, can take place more than once and at one or more different locations where the payload system resides. Accordingly, the payload delivery system of the present invention is able to guarantee delivery of the entire payload.

Further, the end-to-end ubiquitous payload delivery system of the present invention can be used with value-added services, that is, standardized communication services such as Directory Services, Business/Accounting Services, Security Services, Compression Services, and Language Services. Directory Services, as delineated in X.500 series of ITU Standards, provides transparent address translation services to the users. Business/Accounting Services provides for the recording of the details of the delivery that are subsequently used for system analysis, upgrades, and the generation of billing for the services rendered. Security Services include, but are not limited to, providing encryption, authentication, and validation service to the users. Compression services provide compression appropriate to the payload (e.g., voice compression, video compression, or data compression), enabling efficiencies in storage and transmission. Lastly, Language Services provide conversion of the payload from one language to another.

In architecture, the payload delivery system of the present invention is configured in computer-based hardware comprising one of the possible locations identified above, though preferably implemented in a desktop workstation computer or a computer-based server. Regardless, the hardware suitable for implementing the present invention includes the appropriate network interfaces, support system interfaces (if a server), a central processing unit (CPU), memory (both random access memory (RAM) and a hard disk), and other necessary interfaces as well known in the industry. The payload delivery system is preferably stored in the memory and includes an input manager and an output manager for processing incoming and outgoing payloads, respectively. For purpose of the present disclosure, a payload can take the form of any digital compilation of data, such as but not limited to a fax, voice mail, paging message, or e-mail (may comprise one or more of the following: text data, image data, video data, audio data, or any combination thereof). Supporting functionality is provided by a copier, a media/protocol converter, a scheduler, an archive, a message store, and a value-added services mechanism. The input manager receives outgoing payloads compiled by the sender in a particular media for delivery to a designated recipient. The input manager checks the payload for validity (e.g., adherence to protocols and error free receipt), checks the address of the payload, and then stores a single copy of the payload in the payload store. It is noted that the payload can be stored in an encrypted/unencrypted, compressed/uncompressed form, as dictated by the system parameters. The input manager also checks to see if any media or protocol conversions or copies are necessary, or if any of the value-added services are required to operate on the payload, and if so, whether this should take place locally or at a remote server or workstation having the payload delivery system. If the conversion is to take place locally, then the input manager sends the payload to the media/protocol converter for conversion, and subsequently to the output manager. Otherwise, the input manager sends the payload to the output manager for delivery. The scheduler works in conjunction with the input and output managers to schedule delivery of messages to take advantage of tariffs, and resource availability.

An application interface is connected to the input manager and the output manager for receiving payloads from and placing payloads on the associated network for delivery. The application interface may contain or interface with a network interface that allows the invention to interface to the appropriate network, examples of which include a local area network (LAN) or an Access Network.

In the case of an incoming payload, the input manager receives the payload, determines if media conversion is necessary, and if so, then sends the payload to the media/protocol converter. The input manager further coordinates the copying of the payload by the copier and the storage of the payload in the payload store. The input manager then sends the payload to the output manager in order to complete delivery of the payload by providing the recipient with a copy, or in order to send the payload on to a next payload delivery system location that is in route to the recipient. The output manager is essentially a slave of the input manager in that the output manager is typically instructed what to do with the payload by the input manager.

In addition, the input manager can send a message to the sender via the output manager to notify the sender whether the delivery was successful, whether the recipient has received the payload, or whether the payload has been sent to the next payload delivery system location in route. Further, the input manager can utilize the output manager to send a message to the sender in order to request the sender for additional instructions if the delivery was not successful, or in order to allow the sender to query the status of a sent payload. The notifications received by the input manager that are destined for another payload delivery system are merely passed to the output manager and sent to the next payload delivery system. The notifications received by the input manager for that payload delivery system are provided to the sender.

It is envisioned that on successful delivery to the recipient, the invention maintains an archived copy of the payload with the delivery details for a duration that is consistent with security, business and service parameters

The present invention can also be conceptualized as providing for a payload delivery method for providing media independent, guaranteed delivery of a payload in accordance with delivery parameters defined by the sender. The payload delivery method can be broadly generalized as follows. Initially, the sender generates a payload for delivery in a first media, for instance, an e-mail or fax. Next, the sender establishes the parameters of the delivery, including the number of attempts within a given period of time, the allowable cost of transmission, the types and/or frequency of notification, etc. The payload is then passed on to the payload delivery system for delivery to the recipient. The payload delivery system can be located at the sender's desktop workstation, the sender's or receiver's server, or a digital switch at either the sender's or receiver's end. The payload delivery system performs the function of converting the entire payload (or, a portion thereof), if necessary, from the first media to an alternate media in order to complete delivery of the payload to the recipient. For example, if the sender prepared an e-mail in a format that is not compatible with the e-mail of the recipient, or could not be delivered as an e-mail, then the present invention converts the sender's e-mail message into a format compatible with the recipient's e-mail, or alternatively, from e-mail to fax. As yet another alternative, the present invention may convert the protocol of the payload to one compatible with the protocol recipients system. Consequently, if the e-mail proves undeliverable for whatever reason, then the e-mail of the sender is converted into an alternate media designated in the delivery parameters so that further attempts at delivery can be made in accordance with the delivery parameters. As a part of the present invention, notifications that the recipient could receive are that the recipient has received the payload, that delivery by a specified media has not been successful, or that a media conversion was performed. The sender, on the other hand, not only may receive the same notifications as those provided to the recipient, the sender is preferably always given notification of delivery so that the sender is guaranteed that the payload has been received by the recipient.

Other features and advantages of the present invention will become apparent to one with skill in the art upon examination of the following drawings and detailed description. It is intended that all such additional features and advantages be included herein within the scope of the present invention, as defined by the claims.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention can be better understood with reference to the following drawings. The elements of the drawings are not necessarily to scale, emphasis instead being placed upon clearly illustrating the principles of the present invention. Furthermore, referenced numerals designate corresponding parts throughout the several views.

FIG. **1** is a block diagram of a communication network incorporating the end-to-end, guaranteed, ubiquitous payload delivery system of the present invention;

FIG. **2** is a block diagram of an illustrative computer system that includes the payload delivery system of the present invention;

FIG. **3** is a block diagram of the architecture and functionality of the payload delivery system of FIG. **2**;

FIG. **4** is an illustrated database entry for a recipient that establishes the delivery parameters of a payload for use with the delivery system of FIG. **2**; and

FIGS. **5**, **6**, **7**, **8**, **9**, and **10** are illustrative examples of payload delivery with the delivery payload system of FIG. **2**.

FIG. **10** is a flow chart illustrating the methodology and functionality of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

The following description is of the best presently contemplated mode of carrying out the present invention. This description is not to be taken in a limiting sense, but is made merely for the purpose of describing the general principles of the invention. Consequently, the scope of the invention should be determined by referencing the appended claims.

I. Architecture

With reference to FIG. **1**, the end-to-end ubiquitous payload delivery system of the present invention provides a payload delivery system and associated methodology that can be employed in connection with a communication network **12** for the purpose of providing guaranteed payload delivery between users in a media independent environment. The payload delivery system is preferably implemented as a computer program for use by or in connection with a computer-based system such as a workstation or a server. As such, the payload delivery system is essentially a computer application that can be stored on any computer-readable medium, such as but not limited to electronic, magnetic, optical, or other physical device or means that can contain or store a computer program for use by or in connection with a workstation or server in the communication network **12**. Accordingly, the payload delivery system is a platform independent application with a versatility of being able to be located in one or more locations within the communication network **12** as described in more detail below.

The communication network **12** comprises a wide area network (WAN) **13**, a local area network (LAN) **38**, and a plurality of subscribers (e.g., **30**, **32**, **36**). The WAN **13** includes a long distance portion **14**, a first local digital switch **24**, and a second digital switch **26**. The long distance network portion **14** enables both domestic and international long distance services via a first long distance digital switch **16** and a second long distance digital switch **18** which are remotely located with respect to one another and interconnected by a transport link **20**. For purposes of the present invention, the transport link **20** may comprise any one of a variety of transport technologies such as fiberoptics, wireless (e.g., radio frequency (RF) or satellite). A long distance server **22** that includes the payload delivery system of the present invention can be connected to either of the digital switches **16**,**18**.

Interconnected by the long distance network portion **14** are the first local digital switch **24** and the second local digital switch **26**. The first and second local digital switches **24**, **26** are those commonly utilize by a local exchange carrier (LEC) for switching in a regional network, as well known in the industry. Therefore, calls originating or terminating within the particular regional network of a local digital switch **24**, **26** are routed by the local digital switch **24**, **26** hosting that regional network to the designated recipient. A local server **28** that includes the payload delivery system of the present invention can be connected to either of the local digital switches **24**, **26**.

Within the regional network of each local digital switch **24**, **26** are a plurality of subscribers such as a computer workstation **30** or a digital phone **32** that are interconnected to the first local switch **24** via respective subscriber lines **34**, or a computer workstation **36** that is interconnected with the second local data switch **26** via the subscriber line **40**

As illustrated in FIG. **1**, the LAN **38** comprises a LAN server **42** that networks a plurality of customer provided equipment (CPE) such as a fax machine **44** and a plurality of work stations **46**. It is worth noting at this point that the interconnections **34**, **40** can be, but are not limited to, fiberoptics, wireless, RF, coax or twisted pair. A local server **31** that includes the payload delivery system of the present invention can be connected to the LAN server **42**.

In accordance with an important feature in the present invention, the payload delivery system of the present invention does not have to be located at any one particular location in the communication network **12**, but may reside in a variety of different locations including at least any one or more of the following: the workstation **46**, the computer workstations **30**, **36**, the LAN server **42**, the local server **31** associated with the LAN **38**, the local servers **28** associated with local switches **24**, **26**, or the long distance servers **22** associated with long distances switches **16**, **18**. In addition, if the transport link **20** comprises a satellite, the payload delivery system of the present invention may reside in the satellite. However, it is preferred that the payload delivery system of the present invention be supported at more than one location in order to provide redundancy and load management capability.

With reference to FIG. **2**, shown is a computer system **50** illustrative of a typical computer architecture found in workstations and servers, and that is suitable for employing the payload delivery system of the present invention. Accordingly, as mentioned above, the computer system **50** that is implementing the payload delivery system of the present invention can be any one or more of the workstation **46**, the LAN server **42**, the local server **31**, the workstations **30**, **36**, the local servers **28**, or the long distance servers **22**. Note, the computer architecture illustrated in FIG. **2** is well known in the art and is provided merely for the purposes of describing the present invention

The computer system **50** comprises a conventional central processing unit (CPU) **52** that communicates to other elements within the computer system **50** via a system interface **54**. The system interface **54** contains both data and control buses which are shown as combined in order to simplify the computer system **50**. The CPU **52** is preferably capable of running processes in order to support the functionality of the

EXHIBIT H
PAGE 14 OF 20

payload delivery system of the present invention. An input device 56, for example, a keyboard or mouse, is used to input data from a user of the computer system 50, and a screen display device 58 is used to output data to the user. A memory 60 within the computer system 50 stores the payload delivery system 62 which communicates with a conventional operating system 64 for the execution of the payload delivery system via the CPU 52. A memory bus 66 is preferably provided to interconnect the CPU 52 and memory 60 in order to allow high speed communication between the operating system 64 and the CPU 52. A support system interface 68 interconnects the computer system 50 to a support system (not shown) in order to provide operation, administration, maintenance, and provisioning capabilities, as is common in a server architecture. A network interface 70 interconnects the computer system 50 to the communication network 12 (FIG. 1). Other standard interfaces, including serial and parallel interfaces, are provided by standard interfaces 72. Lastly, a tape drive 74 and a disk drive 76 are provided for backup, and data and program storage capabilities, respectively.

With reference now to FIG. 3, the payload delivery system 62 comprises an application interface 80, an input manager 82, an output manager 84, a copier 86, a media/protocol converter 88, a scheduler 90, an archive 92, a payload store 94, and a value added services mechanism 96. The application interface 80 interconnects the payload delivery system 62 to the communication network 12 for receiving and sending payloads via the network interface 70. The network interface 70 (FIG. 2) may indeed be a component of the application interface 80. In specific regard to the payloads received by the application interface 80, the application interface 80 checks them for adherence to the protocol and for correctness (e.g., to determine if there is any missing or corrupted information, or to check for adherence to protocols). The payloads received by the application interface 80 are sent to an input manager 82 where they are processed.

The input manager 82 processes payloads, routing them to an appropriate functional module such as the copier 86, the media/protocol converter 88, the scheduler 90, the archive 92, the payload store 94, or the value added services mechanism 96. The input manager 82 is interconnected with each of the aforementioned functional modules via a bus 98.

The copier 86 is provided to copy the payload for enabling distribution of the payload to more than one recipient, as will be discussed in greater detail below.

The media/protocol converter 88 is provided to convert the media of the payload from its originating media to one or more alternate media as designated by the sender in the delivery parameters or as otherwise necessary to complete delivery. It should be noted that this can entail the conversion of the entire payload, or only a portion of the payload, in order to effectuate delivery. In addition, the media/protocol converter 88 converts the particular protocol of a payload as necessary when the payload delivery system 62 detects an incompatibility between protocols. It should also be noted that media and/or protocol conversion can take place pre- and/or post-delivery, and at any one or more of the locations where the payload delivery system resides.

The scheduler 90 manages the internal processing of payloads in the payload delivery system 62. For example, the scheduler 90 may, subject to the parameters specified by the sender, schedule a message to take advantage of tariff structure to the benefit of the sender.

The archive 92 provides a mechanism for storing payloads for extended periods of time, at least more than the period of time required for delivery of the payload.

The payload store 94 provides a mechanism for storing the payload during transmission and/or reception as necessary in the delivery of the payload to the recipient. This will become more apparent upon the following discussion regarding the operation of the payload delivery system 62.

The value added services mechanism 96 is included to provide one or more standardized communication services such as Directory Services, Business/Accounting Services, Security Services, Compression Services, or Language Services. The Directory Services, as delineated in X.500 series of ITU Standards, enables transparent address translation services to the users. The Business/Accounting Services provides for the recording of the details of the delivery that are subsequently used for system analysis, upgrades, and the generation of billing for the services rendered. The Security Services include, but are not limited, to providing encryption, authentication, and/or validation services for the users. The Compression services provide compression appropriate to the payload (e.g., voice compression, video compression, or data compression), enabling efficiencies in storage and transmission. Lastly, the Language Services provide for the conversion of the payload from one language to another

The output manager 84 processes all outgoing messages including payloads and notices to the sender, and sends the same to the application interface 80.

II. Operation

The payload delivery system of the present invention transfers the payload using multiple communication methods, following the rules established by the user for transmission via the delivery parameters. The delivery parameters are preferably set forth in a payload entry associated with the recipient, such as the payload entry 102 illustrated in FIG. 4. It is preferred that the user establish a database of payload entries 102, specifying the applicable communication methods to be used when communicating with the respective recipients, for instance, e-mail, fax, postal, address, telephone number, etc , and a preferred method of communication with the appropriate backup methods (i.e., a secondary media with parameters, a tertiary media and parameters, etc.). For delivery to a postal address, the user specifies a specific carrier such as United States Postal Service, United Parcel Service (UPS), or Federal Express. Thus, once the user has generated a particular payload, the communication can be sent by merely selecting the name of the recipient from the user's database, as is presently done with address books associated with most e-mail applications. As shown in FIG. 4, the name, e-mail address, fax number, fax-print number, and post office address are provided for the delivery parameters by the payload entry 102. Note that this information is recipient specific.

In addition to the recipient specific information provided by the entry 102, payload delivery specific information is also provided, such as the number of retries before a media change, the intervals between media retries, a secondary media, a number of retries with the secondary media, the intervals between retries with the secondary media, a tertiary media, a number of retries with the tertiary media, and the intervals between retries with the tertiary media, and so forth and so on. The aforementioned information comprises the delivery parameters defined by the sender. This is an important aspect of the present invention in that the sender is given the ability to set the delivery parameters of the payload which the sender typically compiled and/or is fully knowledgeable of the contents. Moreover, the sender is usually the one who bears the cost of delivery. Thus, the sender is the

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best person to set the delivery parameters. Other delivery parameters (not shown) can include, for example, the time for delayed delivery or the tariff to be used (media specific). Yet another delivery parameter that is not incorporated into the entry **102** is the capability of allowing the user to group recipients for delivery of a single payload to multiple recipients.

The capability of having the sender set the delivery parameters is particularly significant in the situations where the payload includes data in more than one format, for instance, an e-mail with a first portion compiled in text and a second portion compiled in a spread sheet. In this situation, the e-mail may not be suitable for conversion into a voice message via character recognition because the spread sheet would be lost. However, with the present invention, the sender is aware of this and would most likely designate a secondary media such as fax that would be compatible with both the text and spreadsheet portions of the e-mail. Accordingly, the present invention is able to guarantee delivery of the entire payload, not just that portion compatible with the recipient's system. Alternatively, the payload delivery system with the present invention may be configured to convert the second portion (i.e., the spread sheet) of the e-mail into a fax and the first portion of the e-mail into a voice message (or some other media). Thus, the present invention is able to guarantee delivery of the entire payload to the recipient by matching the available media of the recipient to that which is compatible with the payload.

As illustrative examples of payload delivery with the payload delivery system **62** of the present invention, the following discussion addresses four delivery scenarios as illustrated in FIGS. **5**–**9**. For reasons of brevity, it is assumed that the sender and recipient are both operating from stand alone workstations **30**, **36** (FIG. **1**), and that the payload delivery system **62** is operating on both workstations **30**, **36** and on both of the local servers **28** (FIG. **1**) that are connected to the local digital switches **24**, **26**.

In general, the user initially creates a message for delivery, including filling out a payload entry **102** (FIG. **4**) having the delivery parameters for that payload. The user then sends the message. The message is received by the output manager where it is classified with other outgoing messages based upon the method of delivery. If the message is for delivery to a postal address, based on the carrier specified in the recipient's address, the output manager determines the carrer's point of presence (POP) and proximity to the recipient. Then an e-mail or fax is sent to this point of presence where manual delivery to the recipient is effectuated. In the case of manual delivery, a sender is notified when the delivery is done, not when the carrier receives the e-mail or fax. The sender may not have to know the details of the carrier (e.g., address, fax number, e-mail address), relying on the value added services mechanism **96** to provide this information, based on the address(es) of the recipient. In order to address the carrier concerns and ensure that the payload is not divulged in route, the service requires a specific type of secure form to be used. This is preferably a two-page, sealed form that allows the text to appear on the inside while the details of the recipient (e.g., address, phone number) are provided in the appropriate place on the top of the first page. In order to read the payload inside, a seal has to be broken. In addition, appropriate handshakes between the POP and the delivery manager would ensure that duplicate messages can not be created.

Regardless of the method of delivery, the response received by the input manager regarding the delivery of the message is provided to the sender. Furthermore, the user can at anytime query the output manager to determine the status of the delivery. In addition to notifying the user that the message was delivered to the recipient, it may be preferred at times to notify the user of the methods used other than that of the native media. The method of providing the notification that is performed by the output manager is dependent on the payload-transport technology. As an example, the notification may be achieved with messages on the D-channel of integrated service digital network (ISDN), with little to no overhead in order to provide a great utility value to the end user. With a technology such as wireless, the notification can be achieved by out-of-band signaling, while in other technologies the notification can be achieved with in-band signaling.

With reference to FIGS. **5** and **6**, shown are two instances of delivery for a single-recipient case. In FIG. **5**, the payload delivery system **62** receives a near-end acknowledgment **120** when the near-end server **28** receives the payload, a far-end acknowledgment **122** when the payload is delivered to the far-end server, and a far-end acknowledgment of delivery **124** finally when the end-user server receives the payload. At the near-end, the payload delivery system **62** makes an intelligent choice of whether the message should be replicated for multiple recipients, thereby managing the bandwidth of the backbone network.

In FIG. **6**, the payload delivery system **62** provides the user with a near-end acknowledgment **120'** when the near-end server receives the payload, and a far-end acknowledgment of delivery **124'** when the end-user receives the payload, without the intermediate server-server specific acknowledgments. As a part of the present invention, the user is be able to select the notification method, i.e., either that shown in FIGS. **5** or **6**.

With reference to FIG. **7**, shown is an instance where multiple attempts are made in a manner that is transparent to the user in order to complete delivery. As shown, following an unsuccessful delivery attempt **126** initially, retries **128**, **130** are made before delivery is completed by attempt **130**. Following delivery, a far-end acknowledgment of delivery **132** is received by the user.

With reference to FIG. **8**, shown is an instance where multiple attempts are made before a media conversion is made at the far-end server **28**, in accordance with the delivery parameters defined by the sender. As shown, following an unsuccessful delivery attempts **136**, **138**, notification **140** is received by the user. The payload delivery system **62** responds to the notification **140** by sending a new media notification **142** which results in the far-end server **28** performing a media conversion. Subsequently, delivery with the new media is completed with attempt **144** and a far-end acknowledgment of delivery **146** is received by the user.

With reference to FIG. **9**, shown is an instance where multiple attempts are made before a media conversion is made at the sender's workstation **30**, in accordance with the delivery parameters defined by the sender. As shown, following an unsuccessful delivery attempts **148**, **150**, notification **152** is received by the user. In this case, the payload delivery system **62** responds by performing a media conversion at workstation **30** and retansmitting the payload in the new media via attempt **154** which is then successfully delivered. The user subsequently receives a far-end acknowledgment of delivery **156**.

The multi-recipient case is an extension of the single-extension case with the required extension of payload duplication. The functionality that is required to realize this concept is distributed across the client and the server. More particularly, this exact functionality depends on the factors of optimization (i.e., network bandwidth, cost of client, server, delays, security, etc.).

It should be noted that the methodology and functionality of the present invention is illustrated in the flow chart of FIG. 10.

In concluding the detailed description, it should be noted that it will be obvious to those skilled in the art that many variations and modifications may be made to the preferred embodiment without substantially departing from the principles of the invention. All such variations and modifications are intended to be included herein within the scope of the present invention, as set forth in the following claims. Further, in the claims hereafter, the structures, materials, acts, equivalent of all means or step plus function elements are intended to include any structures, materials, or acts for performing the recited functions in combination with other claimed elements as specifically claimed.

Wherefore, the following is claimed:

1. A payload delivery system for providing guaranteed end-to-end delivery of a payload from a sender to a recipient, said payload being delivered via one or more communication networks, said system comprising:

means for generating a payload in a first media;

means for defining payload delivery parameters by said sender;

means for converting said payload to an alternative media at different locations as necessary for completion of delivery over one of said communication networks; and

means for automatically notifying said sender upon receipt of said payload by said recipient.

2. The payload delivery system of claim 1, further comprising second means for converting said payload to said alternative media, said second means for conversion being remotely located with respect to said first means for conversion.

3. The payload delivery system of claim 1, further comprising means for administering media conversion from said first media to said alternative media for completion of delivery of said payload to said recipient.

4. The payload delivery system of claim 1, wherein said payload delivery parameters include a number of delivery retries before said media conversion to said alternative media.

5 The payload delivery system of claim 1, wherein said payload includes an e-mail message.

6 The payload delivery system of claim 1, wherein said payload includes a fax.

7. The payload delivery system of claim 1, wherein said payload includes an audio file.

8. The payload delivery system of claim 1, wherein said payload includes a video file.

9. The payload delivery system of claim 1, wherein said payload includes a paging message.

10. The payload delivery system of claim 1, wherein said payload includes a sealed security letter.

11. The payload delivery system of claim 1, wherein said one of said communication networks is a local area network.

12. The payload delivery system of claim 1, wherein said one of said communication networks is regional network.

13. The payload delivery system of claim 1, wherein said one of said communication networks is an international network

14. The payload delivery system of claim 1, wherein said one of said communication networks is an Internet.

15. The payload delivery system of claim 1, wherein said one of said communication networks is a Information Superhighway.

16. The payload delivery system of claim 1, wherein said one of said communication networks is a paging network.

17. The payload delivery system of claim 1, further including at least one value-added service.

18. The payload delivery system of claim 1, wherein said payload delivery system is located at a workstation of said sender.

19 The payload delivery system of claim 1, wherein said payload delivery system is located in a local area network.

20. The payload delivery system of claim 1, wherein said payload delivery system is located in a wide area network.

21. The payload delivery system of claim 1, wherein said payload delivery system is located at more than one location.

22. The payload delivery system of claim 1, wherein said payload comprises more than one portion with at least two said portions in a different media, and wherein said means for converting said payload is configured to convert respective said portions of said payload into different media independent of one another.

23. A payload delivery method for providing guaranteed end-to-end delivery of a payload from a sender to a recipient, said payload being delivered via one or more communication networks, comprising the steps of:

generating a payload in a first media;

defining payload delivery parameters by said sender,

converting said payload to an alternative media at different locations as necessary for completion of delivery of said payload; and

automatically nosing said sender upon receipt of said payload by said recipient.

24. The method of claim 23, further comprising the step of administering media conversion from said first media to said alternative media for completion of delivery of said payload to said recipient.

25. The method of claim 23, wherein the step of defining payload delivery parameters comprising the step of defining a number of delivery retries before a media conversion to said alternative media.

26. The method of claim 23, wherein the step of converting said payload to said alternative media comprises the step of converting said payload from said first media into an e-mail.

27. The method of claim 23, wherein the step of converting said payload to said alternative media comprising the step of converting said payload from said first media into a fax.

28. The method of claim 23, wherein the step of converting said payload to said alternative media comprising the step of converting said payload from said first media into a sealed security letter.

29. The method of claim 23, further including the step of performing a value added service.

30. The method of claim 23, wherein the step of converting said payload to alternative media results in a notification to said sender of said conversion.

31. The method of claim 23, wherein an unsuccessful delivery results in a notification to said sender.

32. The method of clam 23, wherein said step of notifying said sender includes requesting said sender to send new instructions on delivery methods.

33. The method of claim 23, wherein the sender queries an input manager on a status of said payload delivery.

34. The method of claim 23, wherein said payload comprises more than one portion with at least two said portions

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in a different media, and wherein said step of converting said payload includes converting said portions of said payload independent of one another.

35. The method of claim **23**, wherein said payload delivery parameters includes an ability to group a plurality of recipients of said payload for delivery.

36. A payload delivery method for providing guaranteed end-to-end delivery of a payload from a sender to a recipient, said payload being delivered via one or more communication networks, comprising the steps of:

generating a payload in a first media;

defining payload delivery parameters by said sender;

converting a portion of said payload to an alternative media as necessary for completion of delivery of said payload; and

automatically notching said sender upon receipt of said payload by said recipient.

37. The method of claim **36**, further comprising the step of administering media conversion from said first media to said alternative media for completion of delivery of said payload to said recipient.

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UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 5,826,034
DATED : October 20, 1998
INVENTOR(S) : Albal

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

In the Abstract, the second appearance of "method" is corrected to "media".

In column 6, line 10 – "utilize" is corrected to "utilized".

In column 10, line 59 – "retansmitting" is corrected to "retransmitting".

In column 11, line 62 – "a" is inserted between "is" and "regional".

In column 12, line 2 – "a" is corrected to "an".

In column 12, line 32 – "nosing" is corrected to "notifying".

In column 12, line 39 – "comprising" is corrected to "comprises".

In column 12, line 47 – "comprising" is corrected to "comprises".

In column 12, line 51 – "comprising" is corrected to "comprises".

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 5,826,034

Page 2 of 2

DATED : October 20, 1998

INVENTOR(S) : Albal

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

In column 14, line 4 – "notching" is corrected to "notifying".

Signed and Sealed this

Thirteenth Day of April, 1999

Attest:

Q. TODD DICKINSON

Attesting Officer

Acting Commissioner of Patents and Trademarks