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Attorneys for Plaintiff, Innovative Communications, LLC

UNITED STATES DISTRICT COURT

CENTRAL DISTRICT OF CALIFORNIA

SOUTHERN DIVISION

INNOVATIVE COMMUNICATIONS, LLC,
 	
        Plaintiff,
 	
    v.
 	
LOOP TELECOMMUNICATIONS INTERNATIONAL, INC.,
 	
        Defendant.
	Case No. SACV 12-02024 DOC (RNBx)
	
	PLAINTIFF INNOVATIVE COMMUNICATIONS, LLC FIRST AMENDED COMPLAINT FOR PATENT INFRINGEMENT AND DEMAND FOR JURY TRIAL
	
	JURY TRIAL DEMANDED

Plaintiff Innovative Communications, LLC ("Innovative Communications"), by way of its First Amended Complaint against Defendant Loop Telecommunication International, Inc., ("Defendant") hereby alleges as follows:

NATURE OF THE ACTION

1.   This is an action for patent infringement of U.S. Patent No. 5,436,930; U.S. Patent No. 5,475,691; and U.S. Patent No. 5,475,713, arising under 35 U.S.C. §§ 1, *et seq.*

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THE PARTIES

2.   Plaintiff Innovative Communications is a limited liability company organized under the laws of Delaware with a place of business at 1209 Orange Street, Wilmington, Delaware 19801.

3.   Defendant Loop Telecommunication International, Inc. is a foreign entity who regularly conducts business in the United States and on information and belief has their United States headquarters at 8 Carrick Road, Palm Beach Gardens, Florida 33418.

JURISDICTION AND VENUE

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

5.   This Court has jurisdiction over the subject matter of this action under 28 U.S.C. §§ 1331 and 1338(a).

6.   Upon information and belief, Defendant regularly conducts business in this judicial district and has committed acts of patent infringement in this judicial district including, *inter alia*, selling and offering to sell infringing products and services in this judicial district.

7.   Upon information and belief, Defendant has ongoing and systematic contacts with this judicial district and the United States. Defendant sells and offers to sell products in this judicial district.

8.   Venue is proper in this judicial district under 28 U.S.C. §§ 1391 and 1400(b).

COUNT I – INFRINGEMENT OF U.S. PATENT NO. 5,436,930

9.   Innovative Communications repeats and realleges the allegations of paragraphs 1 through 8 as though fully set forth herein.

10.   On July 25, 1995, United States Patent No. 5,436,930 ("the '930 Patent"), entitled "Simultaneous analog and digital communications with a selection of different signal point constellations based on signal energy," was duly and legally issued by the

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United States Patent and Trademark Office. A true and correct copy of the '930 Patent is attached as Exhibit A to this First Amended Complaint.

11. Innovative Communications is the assignee and owner of the right, title, and interest in and to the '930 Patent, including the right to assert all causes of action arising under said patent and the right to any remedies for infringement of it.

12. In violation of 35 U.S.C. § 271, Defendant has directly infringed, and continues to directly infringe, including under the doctrine of equivalents, the '930 Patent by making, using, selling, and/or offering for sale in the United States, including in this Judicial District, networking products capable of providing technology involving the simultaneous transmission of audio/voice and data signals, without the authority of Innovative Communications.

13. The '930 Accused Products and Services include, but are not limited to Loop Telecommunication International, Inc.'s "Loop-M 0900 Modem" products and Loop Telecommunication International, Inc.'s support services which utilize those products and services.

14. Innovative Communications provided actual notice to Defendant of its infringement of the '930 Patent in a letter sent by certified mail on November 2, 2012.

15. Defendant has had actual knowledge of the '930 Patent and its infringement of the patent since at least the date that it received the November 2, 2012, letter.

16. Upon information and belief, Defendant has committed and continues to commit acts of contributory infringement of the '930 Patent under 35 U.S.C. § 271(c) by selling, offering to sell, and/or importing products including the '930 Accused Services and Products for the '930 Patent, knowing or willfully blind to the fact that these products and services constitute a material part of the invention, were especially made or especially adapted for use in an infringement of the '930 Patent, and have no substantial non-infringing uses.

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17.   In particular, the '930 Accused Services and Products for the '930 Patent constitute a material part of the claimed invention at least because the products include modem devices capable of the simultaneous transmission of audio/voice and data signals. The '930 Accused Services and Products for the '930 Patent were made or especially adapted for use in an infringement of the '930 Patent and have no substantial non-infringing uses at least because they contain components whose only purpose is to implement receiving simultaneous audio/voice and data signals.

18.   Upon information and belief, since at least the date it received the notice letter from Innovative Communications notifying Defendant that the receiving simultaneous audio/voice and data signals feature in its products and services infringed the '930 Patent, Defendant has induced and continues to induce others to infringe the '930 Patent under 35 U.S.C. § 271(b) by, among other things, and with specific intent, actively and knowingly aiding and abetting others to infringe, including, but not limited to, Defendant's customers whose use of the '930 Accused Services and Products constitutes direct infringement of the '930 Patent. In particular, Defendant acted with specific intent to make others, such as its customers, infringe by advertising and selling the products and providing information and/or materials demonstrating infringing uses of the products or services. On information and belief, Defendant engaged in such actions with specific intent to cause infringement, or with willful blindness to the resulting infringement, because Defendant has had actual knowledge of the '930 Patent and its acts were inducing its customers to infringe the '930 Patent since at least the date it received the notice letter from Innovative Communications notifying Defendant that the receiving function in its products and services infringed the '930 Patent.

19.   Innovative Communications has been harmed by Defendant's infringing activities.

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20.   Innovative Communications has provided notice of infringement of the '930 Patent to Defendant, but Defendant thereafter continued to infringe the patent. On information and belief, Defendant's infringement has been, and continues to be, willful.

COUNT II – INFRINGEMENT OF U.S. PATENT NO. 5,475,691

21.   Innovative Communications repeats and realleges the allegations of paragraphs 1 through 20 as though fully set forth herein.

22.   On December 12, 1995, United States Patent No. 5,475,691 ("the '691 Patent"), entitled "Voice activated date rate change in simultaneous voice and data transmission," was duly and legally issued by the United States Patent and Trademark Office. A true and correct copy of the '691 Patent is attached as Exhibit B to this First Amended Complaint.

23.   Innovative Communications is the assignee and owner of the right, title, and interest in and to the '691 Patent, including the right to assert all causes of action arising under said patent and the right to any remedies for infringement of it.

24.   In violation of 35 U.S.C. § 271, Defendant has directly infringed and continues to directly infringe, including under the doctrine of equivalents, the '691 Patent by making, using, selling, and/or offering for sale in the United States, including in this Judicial District, networking products capable of providing technology involving the simultaneous transmission of audio/voice and data signals, without the authority of Innovative Communications.

25.   The '691 Accused Products and Services include, but are not limited to Loop Telecommunication International, Inc.'s "Loop-M 0900 Modem" products and Loop Telecommunication International, Inc.'s support services which utilize those products and services.

26.   Innovative Communications provided actual notice to Defendant of its infringement of the '691 Patent in a letter sent by certified mail on November 2, 2012.

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27.   Defendant has had actual knowledge of the '691 Patent and its infringement of the patent since at least the date that it received the November 2, 2012, letter.

28.   Upon information and belief, Defendant has committed and continues to commit acts of contributory infringement of the '691 Patent under 35 U.S.C. § 271(c) by selling, offering to sell, and/or importing products including the '691 Accused Services and Products for the '691 Patent, knowing or willfully blind to the fact that these products and services constitute a material part of the invention, were especially made or especially adapted for use in an infringement of the '691 Patent, and have no substantial non-infringing uses.

29.   In particular, the '691 Accused Services and Products for the '691 Patent constitute a material part of the claimed invention at least because the products include modem devices capable of the simultaneous transmission of audio/voice and data signals. The '691 Accused Services and Products for the '691 Patent were made or especially adapted for use in an infringement of the '691 Patent and have no substantial non-infringing uses at least because they contain components whose only purpose is to implement receiving simultaneous audio/voice and data signals.

30.   Upon information and belief, since at least the date it received the notice letter from Innovative Communications notifying Defendant that the receiving simultaneous audio/voice and data signals feature in its products and services infringed the '691 Patent, Defendant has induced and continues to induce others to infringe the '691 Patent under 35 U.S.C. § 271(b) by, among other things, and with specific intent, actively and knowingly aiding and abetting others to infringe, including, but not limited to, Defendant's customers whose use of the '691 Accused Services and Products constitutes direct infringement of at least claim 1 of the '691 Patent. In particular, Defendant acted with specific intent to make others, such as its customers, infringe by advertising and selling the products and providing information and/or materials demonstrating infringing uses of the products or services. On information and belief, Defendant engaged in such actions with specific intent to cause

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infringement, or with willful blindness to the resulting infringement, because Defendant has had actual knowledge of the '691 Patent and its acts were inducing its customers to infringe the '691 Patent since at least the date it received the notice letter from Innovative Communications notifying Defendant that the receiving function in its products and services infringed the '691 Patent.

31. Innovative Communications has been harmed by Defendant's infringing activities.

32. Innovative Communications has provided notice of infringement of the '691 Patent to Defendant, but Defendant thereafter continued to infringe the patent. On information and belief, Defendant's infringement has been and continues to be willful.

COUNT III – INFRINGEMENT OF U.S. PATENT NO. 5,475,713

33. Innovative Communications repeats and realleges the allegations of paragraphs 1 through 32 as though fully set forth herein.

34. On December 12, 1995, United States Patent No. 5,475,713 ("the '713 Patent"), entitled "Shaped signal spaces in a simultaneous voice and data system," was duly and legally issued by the United States Patent and Trademark Office. A true and correct copy of the '713 Patent is attached as Exhibit C to this First Amended Complaint.

35. Innovative Communications is the assignee and owner of the right, title, and interest in and to the '713 Patent, including the right to assert all causes of action arising under said patent and the right to any remedies for infringement of it.

36. In violation of 35 U.S.C. § 271, Defendant has directly infringed and continues to directly infringe, including under the doctrine of equivalents, the '713 Patent by making, using, selling, and/or offering for sale in the United States, including in this Judicial District, networking products capable of providing technology involving the simultaneous transmission of audio/voice and data signals, without the authority of Innovative Communications.

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37.   The '713 Accused Products and Services include, but are not limited to Loop Telecommunication International, Inc.'s "Loop-M 0900 Modem" products and Loop Telecommunication International, Inc.'s support services which utilize those products and services.

38.   Innovative Communications provided actual notice to Defendant of its infringement of the '713 Patent in a letter sent by certified mail on November 2, 2012.

39.   Defendant has had actual knowledge of the '713 Patent and its infringement of the patent since at least the date that it received the November 2, 2012, letter.

40.   Upon information and belief, Defendant has committed and continues to commit acts of contributory infringement of the '713 Patent under 35 U.S.C. § 271(c) by selling, offering to sell, and/or importing products including the '713 Accused Services and Products, knowing or willfully blind to the fact that these products and services constitute a material part of the invention, were especially made or especially adapted for use in an infringement of the '713 Patent, and have no substantial non-infringing uses.

41.   In particular, the '713 Accused Services and Products for the '713 Patent constitute a material part of the claimed invention at least because the products include modem devices capable of the simultaneous transmission of audio/voice and data signals. The '713 Accused Services and Products for the '713 Patent were made or especially adapted for use in an infringement of the '713 Patent and have no substantial non-infringing uses at least because they contain components whose only purpose is to implement receiving simultaneous audio/voice and data signals.

42.   Upon information and belief, since at least the date it received the notice letter from Innovative Communications notifying Defendant that the receiving simultaneous audio/voice and data signals feature in its products and services infringed the '713 Patent, Defendant has induced and continues to induce others to infringe the '713 Patent under 35 U.S.C. § 271(b) by, among other things, and with specific intent, actively and knowingly aiding and abetting others to infringe, including, but not limited to, Defendant's customers

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whose use of the '713 Accused Services and Products constitutes direct infringement of at least claim 1 of the '713 Patent. In particular, Defendant acted with specific intent to make others, such as its customers, infringe by advertising and selling the products and providing information and/or materials demonstrating infringing uses of the products or services. On information and belief, Defendant engaged in such actions with specific intent to cause infringement, or with willful blindness to the resulting infringement, because Defendant has had actual knowledge of the '713 Patent and its acts were inducing its customers to infringe the '713 Patent since at least the date it received the notice letter from Innovative Communications notifying Defendant that the receiving function in its products and services infringed the '713 Patent.

43.   Innovative Communications has been harmed by Defendant's infringing activities.

44.   Innovative Communications has provided notice of infringement of the '713 Patent to Defendant, but Defendant thereafter continued to infringe the patent.   On information and belief, Defendant's infringement has been and continues to be willful.

JURY DEMAND

Pursuant to Rule 38 of the Federal Rules of Civil Procedure, Innovative Communications demands a trial by jury on all issues triable as such.

PRAYER FOR RELIEF

WHEREFORE, Innovative Communications respectfully demands judgment for itself and against Defendant as follows:

A.   An adjudication that Defendant has infringed U.S. Patent No. 5,436,930; U.S. Patent No. 5,475,691; and U.S. Patent No. 5,475,713 (collectively "the IC Patents").

B.   An award of damages to be paid by Defendant adequate to compensate Innovative Communications for their past infringement of the IC Patents and any continuing or future infringement of the IC Patents through the date such judgment is

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1 entered, including pre-judgment and post-judgment interest, costs and expenses as justified

2 under 35 U.S.C. § 284;

3   C.   An injunction ordering Defendant to pay an ongoing royalty in an amount to

4 be determined for any continued infringement after the date judgment is entered;

5   D.   A declaration that this case is exceptional under 35 U.S.C. § 285, and an

6 award of Innovative Communications' reasonable attorneys' fees;
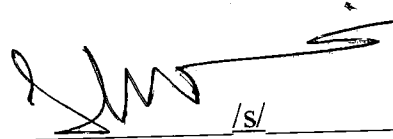
7   E.   An award of treble damages under 35 U.S.C. § 284; and

8   F.   Such other and further relief at law or in equity as the Court deems just and

9 proper.

10   Dated: December 13, 2012

11   LEE, JORGENSEN, PYLE & KEWALRAMANI, PC

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15   By: _____/s/_____

16   H. H. (Shashi) Kewalramani
                            Attorneys for Plaintiff
17   Innovative Communications, LLC

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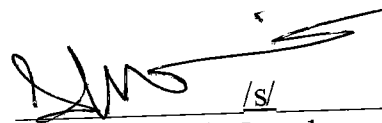21

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DEMAND FOR JURY TRIAL

Plaintiff, Innovative Communications, LLC., hereby demands trial by jury in this action.

Dated: December 13, 2012

LEE, JORGENSEN, PYLE & KEWALRAMANI, PC

By: _____ /s/ _____

H. H. (Shashi) Kewalramani
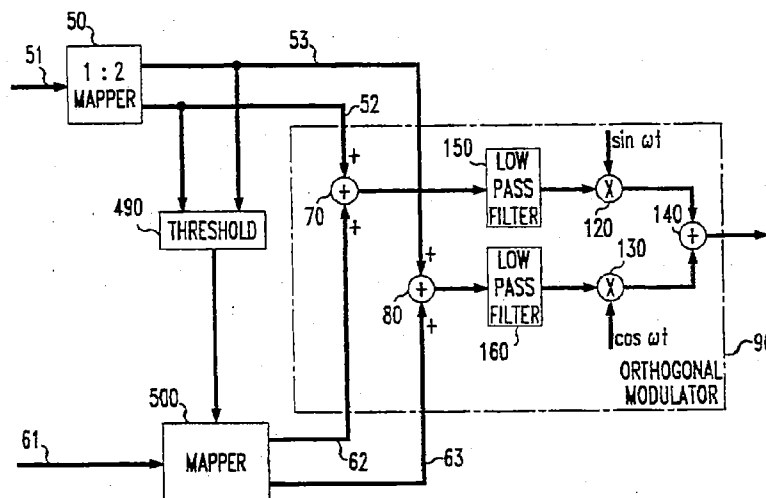Attorneys for Plaintiff
Innovative Communications, LLC

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EXHIBIT A

US005436930A

United States Patent [19]

Bremer et al.

[11] Patent Number: 5,436,930

[45] Date of Patent: Jul. 25, 1995

[54] **SIMULTANEOUS ANALOG AND DIGITAL COMMUNICATIONS WITH A SELECTION OF DIFFERENT SIGNAL POINT CONSTELLATIONS BASED ON SIGNAL ENERGY**

[75] Inventors: **Gordon Bremer; Kenneth D. Ko**, both of Clearwater; **Luke J. Smithwick**, New Port Richey, all of Fla.

[73] Assignee: **AT&T Corp.**, Murray Hill, N.J.

[21] Appl. No.: **76,516**

[22] Filed: **Jun. 14, 1993**

[51] Int. Cl.6 .... H04L 27/04
[52] U.S. Cl. 375/295; 370/110.4; 455/93
[58] Field of Search 375/37, 38, 39, 5, 41, 375/42, 52, 59, 67; 455/74, 93, 102; 370/18, 19, 20, 110.1, 110.4

[56] **References Cited**

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Primary Examiner—Stephen Chin
Assistant Examiner—Hai H. Phan
Attorney, Agent, or Firm—Christopher N. Malvone

[57] **ABSTRACT**

An analog signal is added to a data signal that specifies a symbol in a symbol constellation to produce a combined signal that is transmitted to a receiver. The symbol constellation is picked from a plurality of symbol constellations having differing symbol densities based on the amplitude of the analog signal.

22 Claims, 9 Drawing Sheets

FIG. 1
PRIOR ART

FIG. 2
PRIOR ART
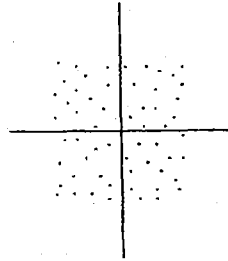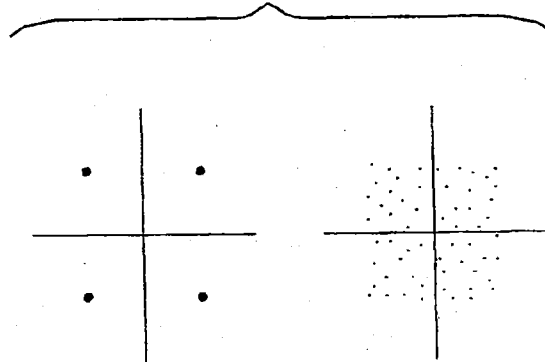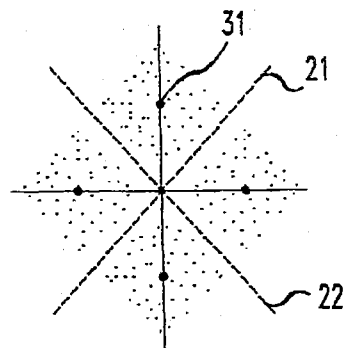
FIG. 3

FIG. 4

FIG. 5

FIG. 6

FIG. 7

FIG. 8

SPECTRUM OF SIGNAL 52 ADDED TO SIGNAL 62

SPECTRUM OF SIGNAL 53 ADDED TO SIGNAL 63

FIG. 9

FIG. 10

FIG. 11

FIG. 12

FIG. 13

PHONE LINE

ANALOG SIGNAL → MAPPER

DIGITAL SIGNAL → MAPPER

+ → CARRIER FREQUENCY ROTATOR

Tx HILBERT PASSBAND FILTER

Tx HILBERT PASSBAND FILTER

ECHO CANCELLER

+

ANALOG HYBRID

+

DIGITAL SIGNAL

SIGNAL SPACE DECODER

CARRIER PHASE COLLECTOR

EQUALIZER

CARRIER FREQUENCY ROTATOR

Rx HILBERT PASSBAND FILTER

Rx HILBERT PASSBAND FILTER

ANALOG SIGNAL → DE-MAPPER

+

+

FIG. 14

REGISTER   55

SCRAMBLING NETWORK   56

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REGISTER

57   REGISTER
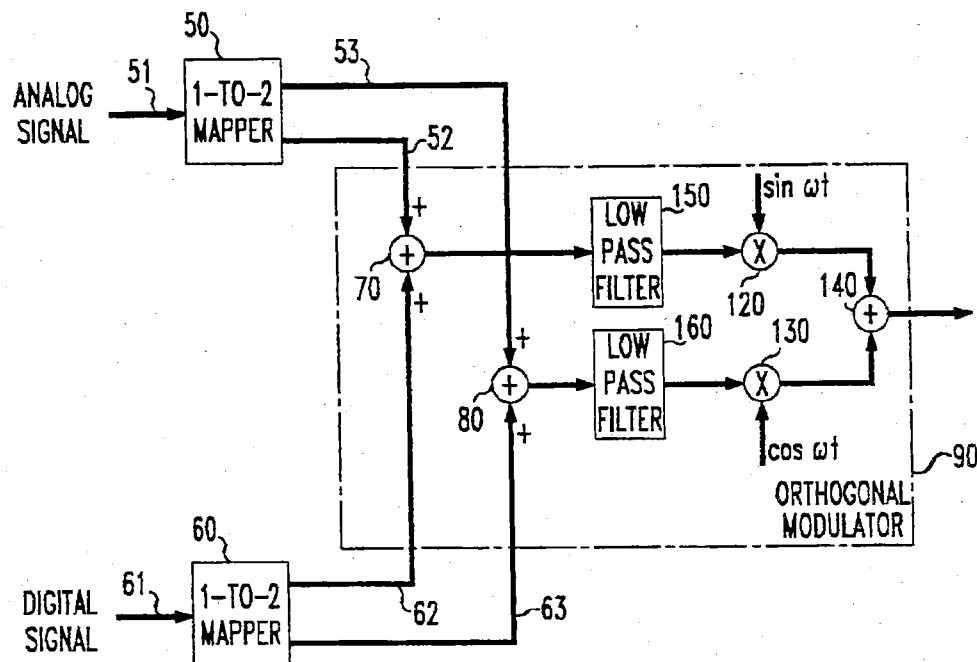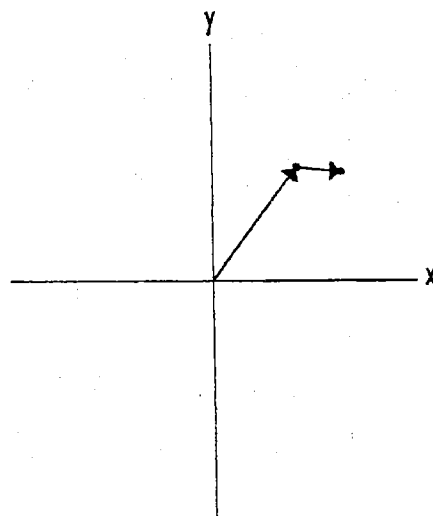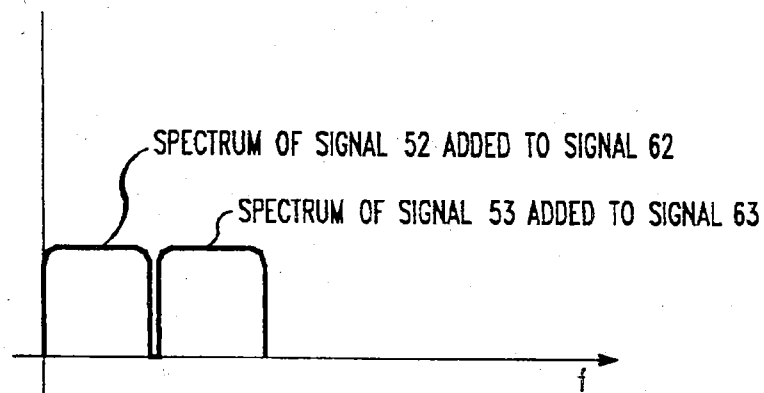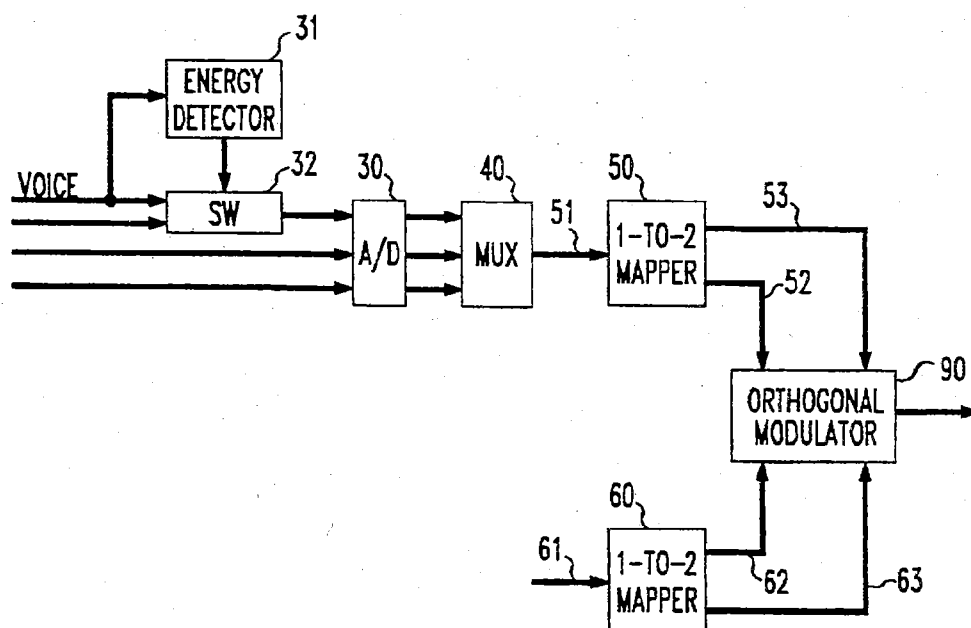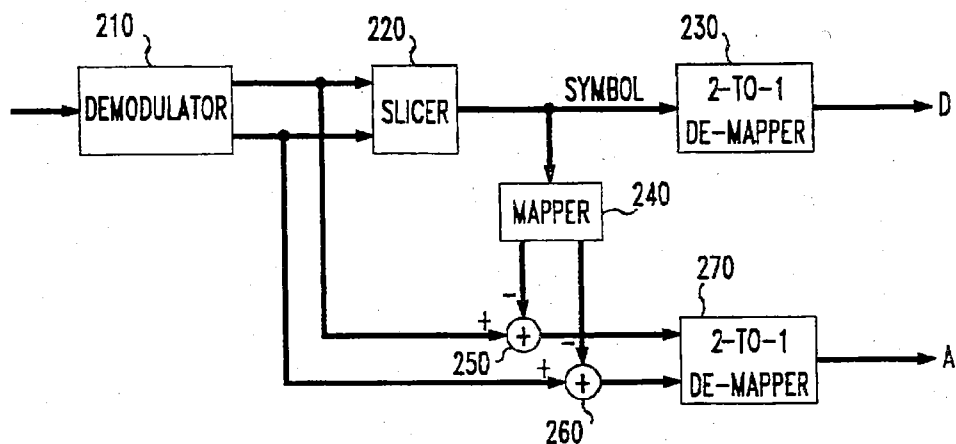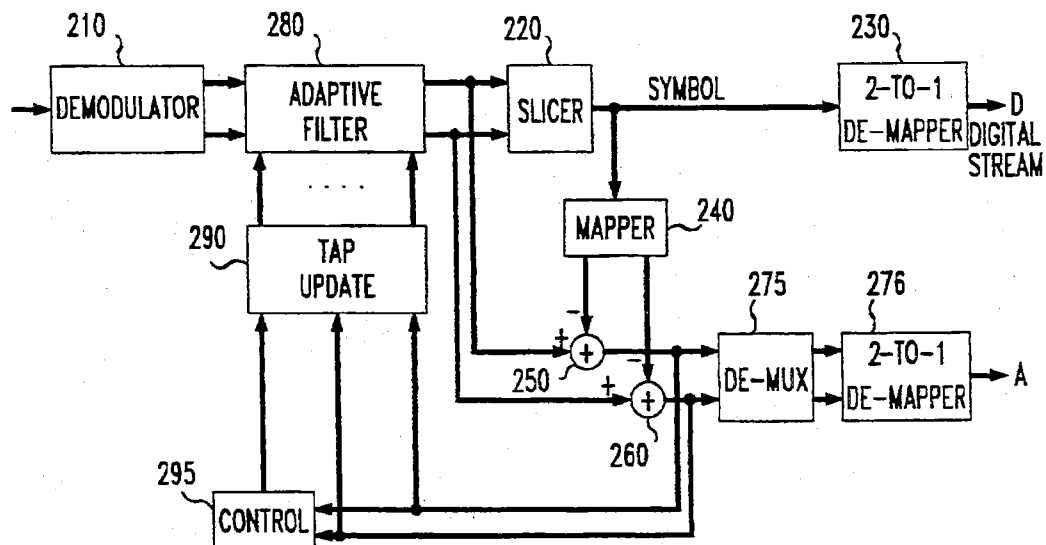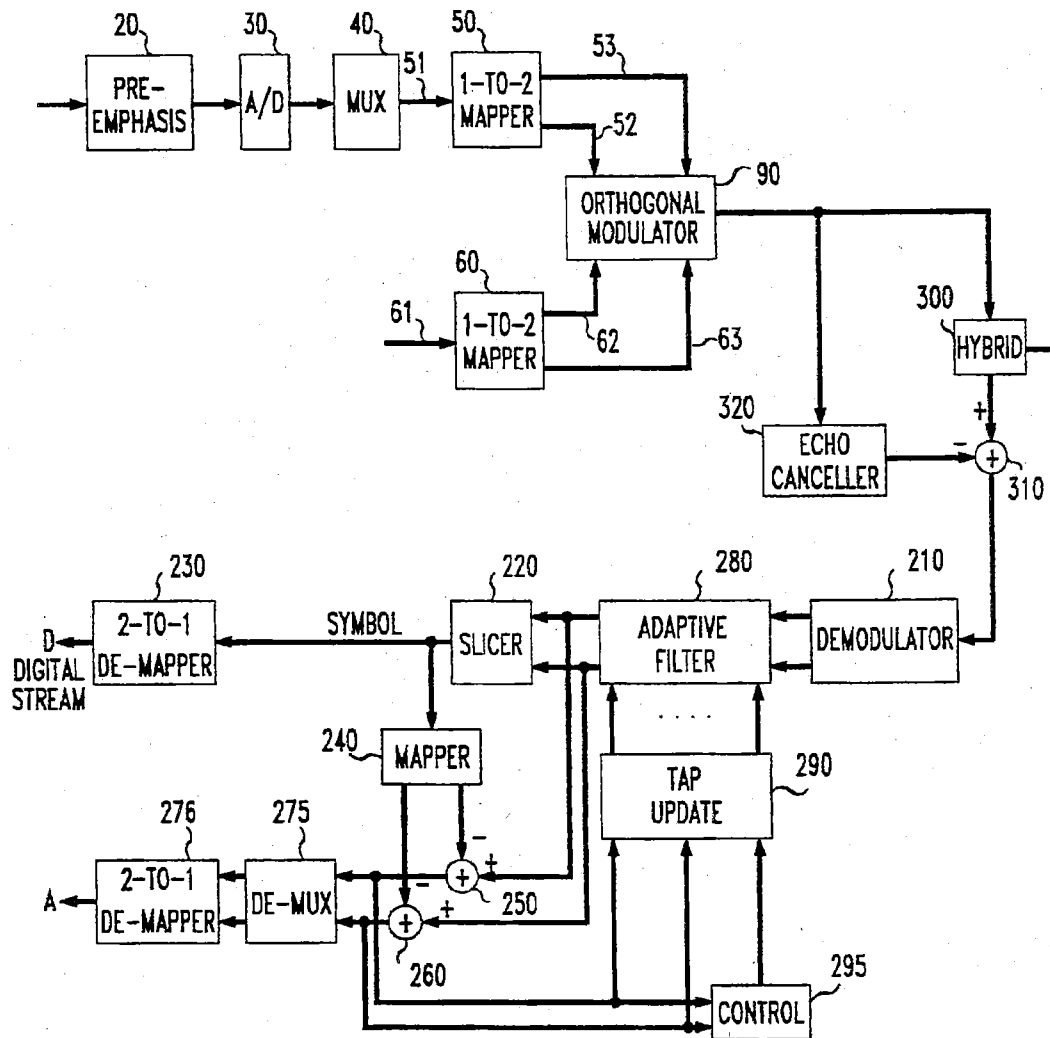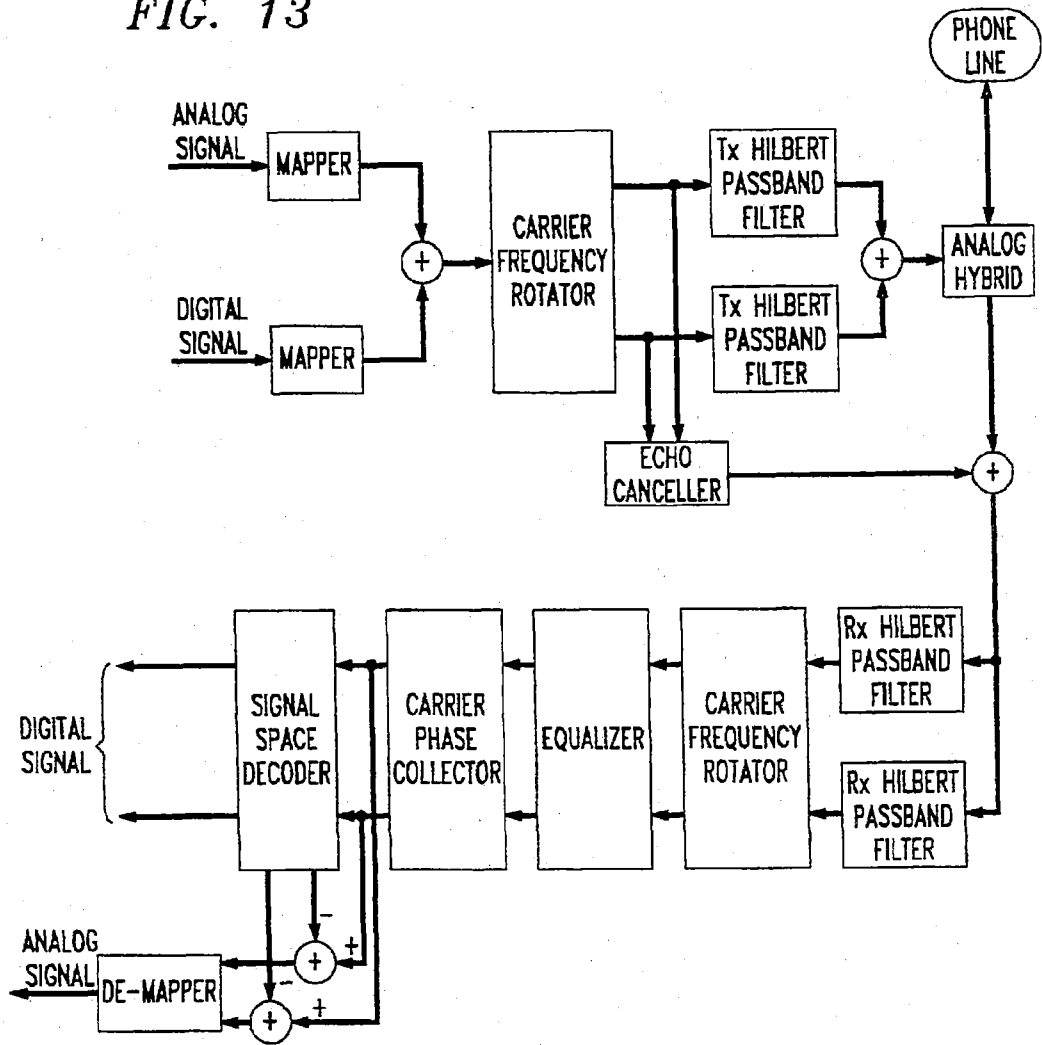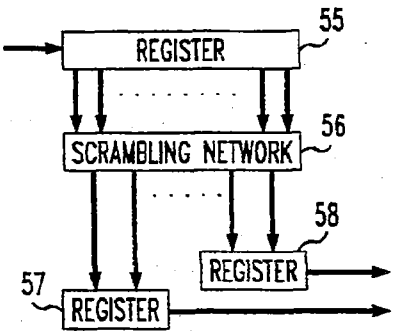
FIG. 15
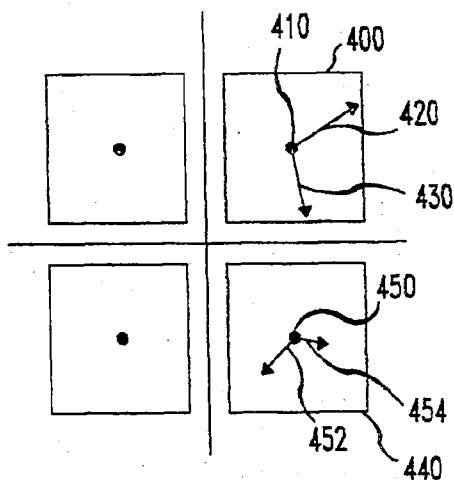
FIG. 16
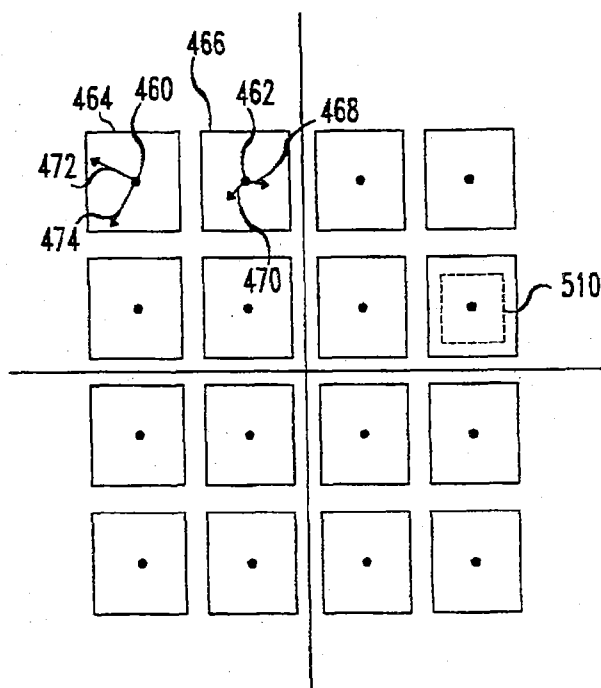
FIG. 17
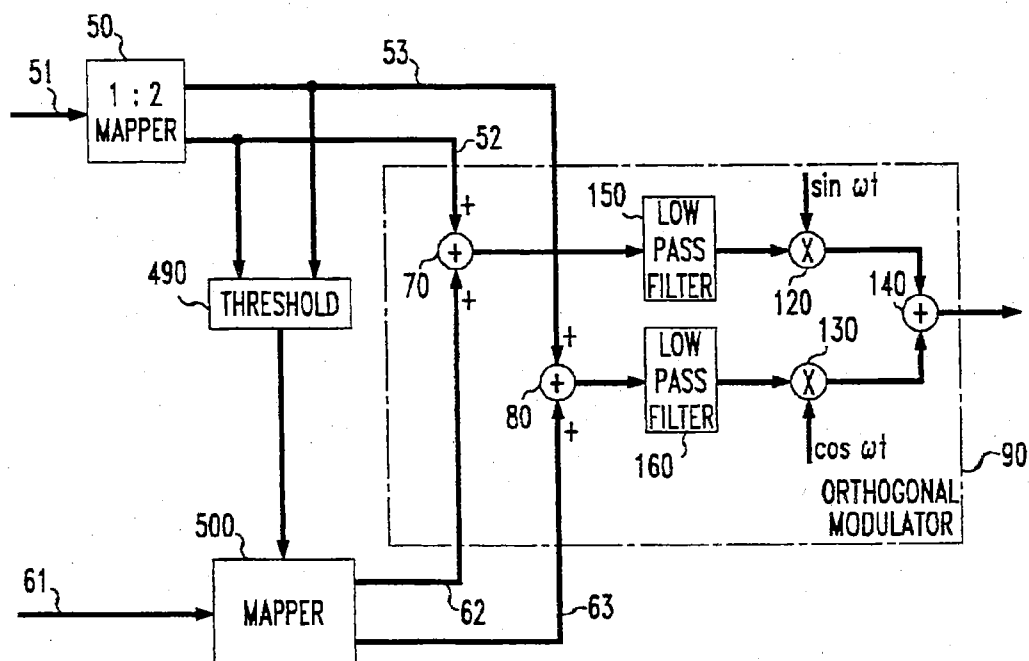
5,436,930

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SIMULTANEOUS ANALOG AND DIGITAL COMMUNICATIONS WITH A SELECTION OF DIFFERENT SIGNAL POINT CONSTELLATIONS BASED ON SIGNAL ENERGY

CROSS-REFERENCE TO RELATED APPLICATION

Related subject matter is disclosed in the application filed concurrently herewith and assigned to the same assignee hereof entitled "Simultaneous Analog and Digital Communication", inventors Gordon Bremer and Kenneth D. Ko.

FIELD OF THE INVENTION

This invention relates to simultaneous transmission of analog and digital signals and, more particularly, to simultaneous transmission of analog signals and digital signals in a non-multiplexed manner and in a co-extensive frequency band.

DESCRIPTION OF THE PRIOR ART

In the prior art, when voice and data is transmitted simultaneously over a channel, it is typically transmitted either via frequency-division multiplexing or time-division multiplexing. In frequency-division multiplexing, the data channel and the voice channel are allocated different sub-bands of the channel's bandwidth. Examples of that are U.S. Pat. No. 4,757,495, U.S. Pat. No. 4,672,602, and U.S. Pat. No. 4,546,212. In time-division multiplexing arrangements, voice signals are sampled, digitized and interleaved with digital data to form a single information stream which is communicated over the available channel. Practically every digital carrier system (e.g. the T1 carrier system) is an example of that.

U.S. Pat. No. 4,512,013, issued Apr. 16, 1985, presents an interesting approach that is close to a frequency division multiplexing arrangement for simultaneous voice and data. The arrangement filters the speech signal and adds thereto a modulated narrowband signal to form the transmitted signal. The narrowband modulated signal derives from a narrowband digital input signal that is modulated with a carrier, thereby shifting the narrow-band up in frequency to a position in the spectrum where there is little speech energy. At the receiver, in reliance of the fact that the speech power is low in the narrowband occupied by the modulated digital signal, the digital signal is recovered through appropriate demodulation. Thereafter, the recovered digital signal is remodulated to replicate the transmitter's operation, adaptively filtered to account for channel characteristics, and subtracted from the received signal. The result is the received speech. As indicated above, one salient characteristic of that arrangement, as stated in col. 2, lines 13–18, is that ". . . an entire analog speech signal and a modulated data signal are capable of being transmitted over a normal analog channel by the multiplexing of the data signal within the portion of the normal analog speech signal frequency band where the speech signal is present and the power density characteristic thereof is low". As an aside, the 4,512,013 arrangement is half duplex.

In the modem art, digital information is communicated over a channel by converting the digital information to analog form. In the most basic form, a modem filters the digital signal (i.e., shifts it in frequency) to form a band-limited signal and modulates that signal to

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reside within the passband of the communication channel. In telephony, for example, that passband may be between 300 Hz and 3500 Hz. To increase the information-carrying capacity of the modulated signal, more sophisticated modems employ quadrature modulation. Quadrature modulation is often depicted as a two-dimensional signal space. Use of the signal space to send voice information is disclosed in U.S. Pat. No. 5,081,647 issued Jan. 14, 1992.

Use of the signal space to send data and voice in described is "High Speed Digital and Analog Parallel Transmission Technique Over Single Telephone Channel", Ajashi et al, IEEE Transactions on Communications, Vol. 30, No. 5, May, 1982, pp. 1213–1218. Unlike prior techniques, where analog and data were segregated into different time slots (TDM) or different frequency bands (FDM), they describe separating analog and data signals into the two different channels of the QAM system. That is, Ajashi et al suggest modulating the in-phase channel with the analog signal, and modulating the quadrature channel with the data signal. Building on that description and concerning themselves with channel equalization, Lim et al analyze equalizer performance in "Adaptive Equalization and Phase Tracking For Simultaneous Analog/Digital Data Transmission", BSTJ, Vol. 60 No. 9, November 1981, pp. 2039–2063. (The 1981 BSTJ article cites the information of 1982 IEEE article as "unpublished work").

No one has achieved the ability to simultaneously sent both data and voice through both channels of a QAM system, and no one has achieved the ability to communicate both by data and analog, simultaneously, and in full-duplex, over a single bidirectional bandlimited communications channel.

SUMMARY OF THE INVENTION

In a communication system that simultaneously transmits analog signals and data signals, it is desirable to identify periods of near silence so that the data rate can be increased to take advantage of the near silence period.

The present invention varies the density of a symbol or signal point constellation used to send data based on the energy of an analog signal that is added to the data signal. When the amplitude of analog signal falls below a threshold, the density of the symbol constellation is increased to take advantage of a lower amplitude analog signal that is added to the data signal.

BRIEF DESCRIPTION OF THE DRAWING

FIG. **1** presents the basic structure of a prior art modem;

FIG. **2** shows the signal space and an illustrative signal constellation for the FIG. **1** system;

FIG. **3** shows the signal space of a QAM analog system;

FIG. **4** shows the signal space of an alternating digital and analog system;

FIG. **5** shows the signal space of a combined digital and analog system;

FIG. **6** presents one embodiment of a transmitter section for a combined digital and analog system;

FIG. **7** presents one orthogonal modulation approach;

FIG. **8** depicts the vector addition that forms the signal space of FIG. **5**;

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FIG. 9 illustrates the arrangements that permit more than one analog signal source to be transmitted simultaneously;

FIG. 10 details the major elements in a receiver in accordance with the principles of this invention;

FIG. 11 presents a block diagram of a receiver that includes adaptive equalization;

FIG. 12 presents the block diagram of an entire modem;

FIG. 13 presents a slightly different embodiment of the FIG. 12 modem;

FIG. 14 depicts one structure for scrambling analog samples;

FIG. 15 illustrates a four-symbol or signal point constellation where the regions around each signal point may be occupied by an analog signal;

FIG. 16 illustrates a higher density symbol or signal point constellation where the regions around each signal point may be occupied by a lower amplitude analog signal; and

FIG. 17 illustrates a block diagram of a transmitter that embodies the present invention.

DETAILED DESCRIPTION

To place this invention in context, FIG. 1 presents a very basic block diagram of a modem that communicates digital data via quadrature modulation techniques. Section 100 is the modem's transmitter section and section 200 is the modem's receiver section. Specifically, in the transmitter section digital data is applied in FIG. 1 to a 1-to-2 mapper 110, and mapper 110 develops two outputs which typically are referred to as the in-phase and quadrature samples. The in-phase samples are applied via low pass filter 150 to modulator 120, which multiplies the applied signal by a carder—i.e, sinωt in FIG. 1. The quadrature samples are applied via low pass filter 160 to modulator 130, which multiplies the applied signal by a second carder. The second carrier is orthogonal to the first carrier; namely, cosωt. Filters 150 and 160 must be bandlimited to no more than ω, in order to avoid aliasing and to at least half the inverse of the output sample rate of mapper 110. The output signals of modulators 120 and 130 are added in element 140 to develop the analog signal of the modem's transmitter section.

In operation, the digital data applied to the FIG. 1 apparatus is a stream of bits. Element 110 views the incoming signal as a stream of symbols that each comprises a preselected number of consecutive bits, and maps each symbol into an in-phase analog sample and a quadrature analog sample.

Practitioners in the art often describe the operations performed in the FIG. 1 apparatus by means of a signal space diagram, such as shown in FIG. 2. The x axis corresponds to one of the carrier signals (e.g., cosωt) and the y axis corresponds to the other carder signal (sinωt). The in-phase and quadrature samples delivered by element 110, in effect, specify a location in the signal space of FIG. 2. Accordingly, the set of possible samples that element 110 can produce corresponds to a set of sample points (i.e., a constellation of points) in the signal space depiction of FIG. 2. A 4-point signal constellation is shown, by way of illustration, in FIG. 2. It is well known, however, that one can create signal point constellations with a larger number of signal points.

To receive signals that were modulated by the FIG. 1 apparatus in accordance with the specific constellation depicted in FIG. 2, one must only identify whether

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the received signal is in the first, second, third or fourth quadrant of the signal space. That means that there exists great latitude in the signals that are received, and any received signal that is still in the correct quadrant is mapped to the correct constellation signal point in that quadrant. Extended to other (and perhaps larger) constellations, the signal space can be divided into regions and the receiver's decision is made with respect to the region in which the received signal is located. We call these regions "neighborhood" regions.

Returning to FIG. 1 and addressing the modem's receiver section, the modulated signal is applied to demodulator 210. Demodulator 210 recovers the in-phase and quadrature components and applies them to slicer 220. Slicer 220 converts the in-phase and quadrature components into symbols and applies the symbols to de-mapper 230. De-mapper 230 maps the symbols into bit streams to form the recovered digital data stream.

Absent any signal degradation (such as due to noise added in the channel) the signal received by demodulator 210 would be precisely the same as the signal sent by adder 140, and a determination of neighborhood regions in which the signal is found (by slicer 220) would be relatively simple and error-free. However, noise that is added to the transmitted signal shifts the received signal in the signal space and modifies the input to slicer 220. Stated in other words, a noise signal that adds to the signal flowing through the communication channel corresponds to a vector signal in the signal space of FIG. 2 that is added to a transmitted sample point. That added vector is of unknown magnitude and unknown phase. Consequently, added noise converts a transmitted signal that corresponds to a point in the signal space into a region in the signal space. This phenomenon is depicted in FIG. 2 by circle 11. Some refer to this circle as a signal space "noise cloud" surrounding the transmitted signal.

From the above it is clear that in order to detect the transmitted signals without errors, the neighborhood regions must be large enough to encompass the noise cloud. Since the average power of the sent signal is typically limited by other considerations, the extent to which the signal constellation covers the infinite space represented by the x and y axes is also limited. This is represented in FIG. 2 by circle 12. The restriction imposed by circle 12, coupled with the restriction on the size of the neighborhood regions that is imposed by noise considerations limits the number of transmitted signal points in the constellation.

As indicated above, it has been observed that in typical modem designs the allowable signal power and the expected fidelity of the channel combine to control the constellation size. Less noisy channels allow for larger constellations, and larger constellations permit higher digital data throughputs. This leads to a totally revolutionary idea of utilizing all, or essentially all, of the available signal space for the transmission of information. A transmitter signal space in accordance with this revolutionary approach is depicted in FIG. 3 where a plurality of signal points are depicted randomly within the signal space. These points are illustrative of the various vectors that the transmitter is allowed to send out. There are no more "constellations of points", where a decision must be made between constellation points; there is only the entirety of the signal space. In other words, rather than having digital signals that are mapped onto a fixed constellation within a signal space, FIG. 3 depicts analog signals that are mapped onto a

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signal space. When the analog signals that form the in-phase component are independent of the analog signals that form the quadrature component, the viable signal space of FIG. 3 may be rectangular.

Having recognized the advantages of sending analog signals in accordance with the signal space of FIG. 3, the next innovation is to alternate between the signal spaces of FIG. 2 and FIG. 3. That is, the innovation is to send customer analog signals or customer digital signals as the need arises. This is depicted in FIG. 4.

Further, having recognized the advantages of sending either analog or digital signals in accordance with the signal spaces of FIG. 4, it was discovered that a totally different communication approach can be taken, that communicating both analog and digital signals, can be expressed concurrently, in a combined signal space. This is illustrated in FIG. 5, where four neighborhoods are identified for illustrative purposes, with demarcation borders identified by dashed lines 21 and 22.

It may be noted that, for purposes of this disclosure, according to the FIG. 5 depiction, the analog signals that form "signal clouds" around each digital constellation point (e.g., point 31) should be restricted in their dynamic range to be totally contained within the neighborhood regions. Hence, here too there is a wade-off between constellation size (which directly affects digital through-put) and dynamic range of the transmitted analog signal (which in some situations translates to "resolution").

FIG. 6 depicts an arrangement that very basically illustrates the principles of this invention. It includes a 1-to-2 dimensional mapper 60 responsive to digital signals applied on line 61. Mapper 60 develops two output signals on lines 62 and 63, each of which possesses pulses with quantized amplitudes that relate to the digital signals arriving on line 61. FIG. 6 also includes a 1-to-2 mapper 50 that responds to an applied analog signal on line 51, and it develops two output signals on lines 52 and 53, each of which possesses pulses with continuous amplitudes that relate to the analog signal on line 5. Outputs 52 and 62 are combined in adder 70 and outputs 53 and 63 are combined in adder 80. The outputs of adders 70 and 80 form the components of the signals that are represented by the signal space of FIG. 5. As in FIG. 1, the outputs of adders 70 and 80 are applied via low pass filters 150 and 160 to modulators 120 and 130 and summed in adder 140 to form a modulated signal as is typically known in the modem art.

In FIG. 6 element 60 is depicted as a 1-to-2 mapper. However, it should be understood that element 60 can be an M-to-N mapper. That is, element 60 can be responsive to a plurality (M) of digital signals and it can develop a different plurality (N) of output signals. Similarly, element 50 can be a J-to-K mapper that is responsive to a plurality of analog signals. Likewise, the collection of elements that follow elements 50 and 60 (i.e., elements 70, 80, 120, 130, 140, 150 and 160), which form orthogonal modulator 90 can be constructed to be responsive to whatever plurality of outputs of that elements 50 and 60 are designed to produce. More specifically, those elements must account for all of the applied input signals, and that means that they must be able to handle K or N signals, whichever is larger. In such a circumstance, however, the user can assume that the larger of the two (K or N) is the dimensionality of the system, and some of the dimensions have either no digital data, or no analog data, whichever applies. Of course, if there are "dimensions" for which there is no

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digital or analog dam, other information can be sent over those dimensions, such as equalization "side" information.

In the context of a signal space, the N pluralities of output signals of elements 50 and 60 (assuming N is larger than K) correspond to the collection of components of vectors in multi-dimensional space; e.g., N-dimensional space. The coordinates of this multi-dimensional space correspond to the orthogonal modulation signals within orthogonal modulator 90. In FIG. 6, the two orthogonal modulation signals are $\cos\omega t$ and $\sin\omega t$, but other modulation signals are also possible; for example, code division multiplexing (CDMA) templates. For purposes of this invention, orthogonal modulation signals are modulation signals that develop a transmitted signal comprising concurrent element signals and yet allow the receiver to separate the received signal into its constituent element signals, those being the signals developed in response to each of the modulation signals. It may also be observed that, relative to FIG. 5, orthogonal modulator 90 performs vector summation of the symbol vector represented by the components developed by element 60 with the analog information vector represented by the components developed by element 50. This is depicted in FIG. 7.

In connection with FIG. 1, it may be noted in passing that the principles of this invention may be utilized even when the output signals of adders 70 and 80 are communicated (e.g., transmitted) directly, without the benefit of combining them in orthogonal modulator 90. Also, orthogonal modulator 90 can simply be a band-shifting means. To the extent that the output of adder 70 (for example) is bandlimited, the output of adder 80 can be shifted beyond the band-limited output signal of adder 70 and combined with the output signal of adder 70. This is presented in FIG. 8. It may also be appreciated that the principles of this invention may be exercised without the use of element 60 in those situations where no digital streams are presented.

To this point in the instant disclosure the implication has been that the input signal applied to element 50 of FIG. 6 is analog. However, that does not have to be strictly the case. In accordance with conventional techniques, an analog signal that is bandlimited can be sampled (within the proper Nyquist bounds). Hence, it should be understood that the input signal to element 50 can be a sequence of analog samples. Moreover, a sampled analog signal can be quantized and represented in digital form. Indeed, an analog signal that has been sampled and converted to digital form can then be converted to amplitude quantized pulse amplitude-modulated format. All of those representations are representations of an analog signal. For example, the collection of the amplitude-quantized PAM pulses is identical to the original analog signal within the bounds of the quantization errors introduced by the sampling and quantizing (A/D conversion followed by D/A conversion) processes.

The fact that sampling and amplitude quantization of the analog signal at the input of element 50 is permitted offers a number of benefits. For one, it allows the signal to be presented to element 50 in digital format. For another, it permits simple multiplexing of different information sources. Thus, for example, elements 50, 60 and 90 can be implemented in accordance with present day modem realizations; i.e., with one or more microprocessors operating under stored program control.

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An example of input signal multiplexing is shown in FIG. 9, which presents an embodiment that includes an A/D converter bank 30 followed by a multiplexer 40. Converter bank 30 converts a plurality of analog signals, such as on lines 33 and 34, to digital format and multiplexer 40 multiplexes its input signals and applies them to element 50. Elements 30 and 40 are conventional A/D and multiplexer elements, respectively.

The combination of elements 30 and 40 allows applying a number of narrowband analog signals to orthogonal modulator 90. The primary limitations are the carrier frequency and the allowable transmission bandwidth of the channel. The narrowband signal can, of course, come from any source. For example, a system installed in an ambulance may sacrifice some voice bandwidth in order to allow narrowband telemetry data of blood pressure and heart pulse rate to be communicated concurrently with the voice.

Additionally, a voice signal energy detector may be included, such as disclosed in U.S. Pat. No. 5,081,647, which would detect periods of silence and send less urgent telemetry data during those silence periods. This is illustrated by elements 31 and 32 in FIG. 9.

The fact that the input to element 50 is digital (in a digital implementation of elements 50, 60 and 90) and that the input to element 60 is also digital should not be confused. The digital input to element 60 is a stream of digits that are each equally important. Hence, those digits are converted into symbols and the symbols into constellation points, and the constellation points are within neighborhoods which are identified by a slicer (e.g., slicer 220 in FIG. 1) within a modem's receiver section. In contradistinction, the digital signals applied to element 50 correspond to digital words that represent amplitude, and the specific interrelationship between adjacent bits of the digital words is maintained. As indicated above, the signal cloud around a signal point within a constellation does not represent a plurality of signal points that must be distinguished, and that is a fundamental distinction.

FIG. 10 presents a basic block diagram of a modem's receiver section in conformance with the principles of this invention. The modulated input signal received from the channel is applied to demodulator 210 which develops the in-phase and quadrature components. Those are applied to slicer 220 which identifies the symbols, and the symbols are applied to de-mapper 230. All this is in accord with conventional modem approaches, as described in connection with FIG. 1. In addition, FIG. 10 includes a mapper 240 that is responsive to the symbols developed by slicer 220. The output of mapper 240 is an accurate estimate of the set of in-phase and quadrature components (that are applied in the FIG. 1 arrangement to elements 150 and 160). The outputs of mapper 240 are subtracted from the outputs of demodulator 210 in subtracters 250 and 260. The outputs of subtracters 250 and 260 are applied to 2-to-1 de-mapper 270 which recombines the analog samples to form an estimate of the original analog signal. De-mapper 270 performs the inverse function of mapper 50.

In may be noted that slicer 220 can be designed to directly provide the output signals that mapper 240 develops; and moreover, de-mapper 230 can be made responsive to such signals. That would alter the FIG. 10 in the sense that slicer 220 and mapper 240 would combine to form a single element and de-mapper 230 as well as adders 250 and 260 would be responsive to that combined element.

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In analog realizations of this invention (e.g., FIG. 6), mapper 50 is responsive to analog signals. Various approaches can be taken to develop the plurality of outputs (two outputs, in the case of element 50 shown in the FIGS.). For example, a single bandlimited analog signal can be divided into a plurality of baseband signals by simply filtering and modulating selected sub-bands. Alternatively, element 50 can accept a plurality of bandlimited analog signals and assign each one of the plurality of bandlimited analog signals to different outputs of element 50.

In time sampled realizations of this invention (whether the realization continues with analog circuitry or digital circuitry), element 50 can simply route alternate samples of a single analog signal to different outputs of element 50, or multiplex a plurality of analog signals and distribute the samples of those signals in any convenient manner.

In order to allow for nonlinear techniques that may be employed to enhance the communication qualities of this invention, it is important to effect equalization of the channel in order to minimize intersymbol interference. Conventional modem technology can be brought to bear to this need.

FIG. 11 presents a block diagram of an arrangement that incorporates equalization. Specifically, FIG. 11 is depicted with a modulator that is followed by equalization hardware (which together can be thought of as a super-demodulator). The equalization hardware comprises an adaptive filter 280 that is interposed between demodulator 210 and slicer 220. The operational characteristics of filter 280 are controlled by filter coefficients that are stored—in modifiable form—within tap update block 290. Tap update block 290 is responsive to the output signals of subtracters 250 and 260. The adaptation of filter 280 is carried out in accordance with conventional modem techniques. The outputs of subtracters 250 and 260 are also applied to demultiplexer 275 and the outputs of demultiplexer 275 are applied to de-mapper 276. De-mapper 276 comprises a bank of de-mappers 270 of FIG. 10. Elements 275 and 276 are included to illustrate a receiver that is adapted for applications where a plurality of analog inputs are multiplexed. Of course, in applications where there is no multiplexing, de-mapper 270 can be substituted.

In accordance with some adaptation approaches, it is easiest to carry out adaptation and the corresponding coefficient updates when the power in the analog signal is small. To limit the process to such intervals, FIG. 11 includes a power detector within control element 295 that is responsive to subtracters 250 and 260. Block 295 is also conventional. It includes a power detection circuit that evaluates the power contained in the signals of subtracters 250 and 260 and delivers a control signal to block 290 to enable (or disable) the coefficient updating process. Of course, block 295 may be more generic, in that the control signal can be derived from other than the analog signal, such as from side information from the transmitter.

FIG. 11 depicts one arrangement for effecting equalization of the transmission channel between a sending modem's transmitter section and a receiving modem's receiver section; to wit, at the receiver's front end, following the demodulator. However, it is well known that equalization can be performed anywhere along the channel, going back even to within a modem's transmitter section.

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FIG. 12 depicts the entire, full duplex, modem constructed in accordance with the depictions of FIGS. 9 and 11. More specifically, a transmitter section (FIG. 9) is coupled with a receiver section (FIG. 11) through hybrid 300 and subtracter 310. Subtracter 310 cooperates with echo canceller 320 in the conventional way to subtract unwanted signals from the signal applied to demodulator 210. For sake of simplicity, echo canceller 320 is shown to connected to the output of orthogonal modulator 90, and in analog embodiments of element 320 this is perfectly satisfactory. However, in digital embodiments it is well known that efficiencies can be realized by having the echo canceller be responsive to the outputs of mapper 60, where the signal rate is much lower. An improvement which incorporates the principles of this invention is shown in FIG. 13. It may be noted that some of the elements in FIG. 13 are designated by different labels; such as "Hilbert passband filter", which corresponds to a modulator, etc. These are circuits that attain the desired results through somewhat different calculations and are well known to persons skilled in the modem art.

The echo cancelling is performed, as in all modems, during a training period, when the far end signal source is silent and the echo canceller is adapted to minimize the output of subtracter 310.

In connection with FIG. 6 it has been disclosed that the input to element 50 can be a sampled analog signal, as well as an unsampled analog signal. It has also been disclosed above that when element 50 is a 1-to-2 mapper (as compared to 1-to-N mapper) and the desired output of element 50 is pairs of a sampled analog signal, the pairs of analog samples can be derived by simply delaying the incoming analog signal by 1/B and sampling both the delayed and the undelayed versions at rate B. This provides sample pairs that correspond to adjacent samples of the original analog signal sampled at rate ½B seconds. Actually, privacy of the communication is enhanced when the samples are not adjacent, and FIG. 14 presents one approach for deriving pairs from non-adjacent samples. It basically includes an input register 55 for storing K analog samples that arrive at rate 2B, a scrambling network 56 that scrambles the outputs of register 55 and develops K outputs, and registers 57 and 58 that are responsive to the outputs of network 56. Registers 57 and 58 store K/2 analog samples every K/2B seconds and output the stored samples at rate ½B seconds. Scrambling network 56 may be simply a cross-connect field.

Other configurations are possible and other elements can be included to enhance operation of the configurations. For example, the "analog" input that enters orthogonal modulator 90 can be filtered to pre-emphasize the high frequencies and, correspondingly, the "analog" output of subtracters 250 and 260 can be filtered to remove the pre-emphasis. The preemphasis can be effected, for example, within the A/D converter 30 or even prior thereto, such as in pre-emphasis filter 20 shown in FIG. 12. The filtering can be done while the "analog" signal is truly analog, or it could be done when the "analog" signal is represented digitally—such as when the transmitter and receiver sections are effected with digital hardware.

FIG. 15 illustrates a four-symbol or signal point constellation that is used when a larger amplitude analog signal is added to the data signal that specifies a symbol. Square region 400 surrounding symbol 410 illustrates the amplitude of an analog signal that can be added to

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data signals specifying symbol 410, without causing a misidentification of the symbol by slicer 220. These regions should be defined so that they do not overlap, and so that the received symbol can be correctly identified by slicer 220. (This is not a strict requirement. See, for example, copending application "Trellis Coding in a Simultaneous Voice and Data System"). Analog signals represented by vectors 420 and 430 illustrate how analog signals with relatively large amplitudes require a low density constellation that provides relatively large regions around the symbols or signal points. Also included in FIG. 15 is region 440 which surrounds symbol 450. In this example, relatively low amplitude analog signals represented by vectors 452 and 454 illustrate that a great deal of space within region 440 is unused by relatively low amplitude analog signals. In an effort to make a more efficient use of the signal space, it is desirable to use a higher density symbol constellation when the analog signals have a low amplitude.

FIG. 16 illustrates a higher density symbol or signal point constellation. The constellation of FIG. 16 includes symbols 460 and 462. Each symbol in the constellation has a surrounding region. The square regions illustrate the amplitude of an analog signal that can be added to a data signal, which specifies the symbol within the region, without causing a misidentification of the symbol by slicer 220. In this illustration, symbols 460 and 462 have regions 464 and 466, respectively. With regard to region 466, low amplitude analog signals added to data signals specifying symbol 462 are represented by vectors 468 and 470. As long as the analog signals have an amplitude that is low enough to keep vectors 468 and 470 within region 466, the high density symbol constellation of FIG. 16 can be used. Vectors 472 and 474 within region 464 illustrate larger amplitude analog signals that were added to data signals specifying symbol 460. When vectors 472 or 474 cross the boundary of region 464, a lower density symbol constellation should be used. The lower density constellation has larger regions around its symbols or signal points and will thereby accommodate larger amplitude analog signals. The larger regions permit slicer 220 to correctly identify a symbol specified by a data signal to which a larger amplitude analog signal has been added.

When the amplitude of the analog signal has increased to a point that will cause the analog signal to extend beyond a region surrounding a signal point, the amplitude of the analog signal may be limited until the mapper switches to a less dense symbol constellation.

It is possible to use more than two types of symbol constellations. For example, when the amplitude of the analog signal is very large, a very low density symbol constellation such as a four-symbol constellation may be used. When the analog amplitude is at an intermediate value, medium density symbol constellations such as 16-point constellations may be used. When the amplitude of the analog signal is very low, much higher density constellations may be used such as 32 or 64 point constellations. Any combination of symbol constellations may be used. Constellations that are less dense or more dense than described herein may be used and still come within the scope of this invention.

The amplitude of analog signal is monitored to determine when the symbol constellation should be changed. This monitoring can be carded out in a variety of ways; however, it is preferable to use a simple thresholding technique.

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FIG. 17 is a block diagram of a transmitter that embodies the present invention. Threshold unit **490** monitors the magnitude of the analog signal produced by encoder or mapper **50**. It is also possible to monitor the magnitude of the analog signal at the input to encoder **50**. Based on the magnitude of the signal from encoder **50**, threshold unit **490** produces an input to mapper **500** that causes the mapper to switch symbol constellations. Threshold unit **490** also supplies a signal that informs the receiver that the symbol constellation will be changed. This information can be passed to the receiver in a variety of ways, for example, threshold unit **490** may inject a digital signal at the input to mapper **500** to indicate an upcoming constellation change. Threshold unit **490** may be fabricated using hardware such as comparators or a software program executed by a microprocessor or microcomputer.

Thresholding can be carded out in a variety of ways. A single threshold may be used to change constellations. For example, when the amplitude of the analog signal falls below a given threshold, a higher density constellation is used, and when the amplitude of the analog signal passes above a threshold, a lower density constellation is used. The threshold should be based on size of the regions surrounding the symbol points of a constellation. For example, the threshold may be set so that it detects an analog signal amplitude that extends beyond a position that is 80% of the distance between a signal point and the boundary of the region surrounding the signal point. In reference to FIG. 16, a threshold of this type is represented by dotted line **510**.

It is also possible to use other thresholding techniques. For example, a higher threshold may be used to determine when to switch from a higher density to a lower density constellation, and a lower threshold may be used to determine when to switch from a lower density to a higher density constellation. This type of thresholding technique can be used to add hysteresis to the process. When multiple constellations are used, several thresholds may be used to determine when to move from a less dense to a more dense constellation, or from a more dense constellation to a less dense constellation.

In another embodiment, one or more characteristics of the analog signal are measured or monitored over a defined period of time, and compared to one or more thresholds. The characteristics monitored can be those of the entire analog signal, or those of one or more orthogonal components of the analog signal. Characteristics such as amplitude, peak amplitude, average amplitude and variations in amplitude may be monitored. After comparing the characteristic(s) to one or more thresholds, a constellation change may be executed immediately, or the constellation change may be limited to occurring at predetermined times. The predetermined times may or may not be related to the periods of time over which signal characteristics are monitored. For example, an average of the analog signal's amplitude may be taken over a period P, and if a threshold comparison indicates that a constellation change is necessary, the change may be executed at a predetermined time after the conclusion of time period P. In another example, the peak analog signal amplitude detected during period P is compared to one or more thresholds, and if a constellation change is necessary, the change is not executed until a predetermined time that is not related to period P.

We claim:

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1. A communication method comprising the steps of:
   monitoring an energy of a first signal;
   choosing a first signal point constellation from a plurality of signal point constellations based on said energy, wherein said plurality of signal point constellations comprises constellations having different signal point densities;
   adding said first signal to a second signal to produce a combined signal, wherein said second signal is one of a plurality of signal points belonging to said first signal point constellation; and transmitting said combined signal.

2. The communication method of claim **1**, wherein said step of monitoring said energy of said first signal comprises monitoring an amplitude of said first signal.

3. The communication method of claim **2**, wherein said step of monitoring comprises taking an average of said amplitude over a predefined period of time and comparing said average with a threshold.

4. The communication method of claim **1**, wherein said step of monitoring said energy comprises monitoring a peak amplitude of said first signal over a predefined period of time.

5. The communication method of claim **1**, wherein said step of monitoring said energy of said first signal comprises comparing an amplitude of said first signal with a threshold.

6. The communication method of claim **1**, wherein said step of choosing said first signal point constellation comprises choosing a constellation having a lower density of signal points when an amplitude of said first signal crosses a first threshold and choosing a second constellation having a higher density of signal points when said amplitude crosses a second threshold.

7. The communication method of claim **1**, wherein said step of choosing said first signal point constellation from said plurality of signal point constellations comprises choosing from at least three constellations.

8. The communication method of claim **1**, wherein said step of monitoring said energy comprises monitoring an orthogonal component of said first signal.

9. The communication method of claim **8**, wherein said step of monitoring said energy of said first signal comprises monitoring an amplitude of said first signal.

10. The communication method of claim **9**, wherein said step of monitoring comprises taking an average of said amplitude over a predefined period of time and comparing said average with a threshold.

11. The communication method of claim **8**, wherein said step of monitoring said energy comprises monitoring a peak amplitude of said first signal over a predefined period of time.

12. The communication method of claim **8**, wherein said step of monitoring said energy of said first signal comprises comparing an amplitude of said first signal with a threshold.

13. The communication method of claim **1**, further comprising the step limiting an occurrence of a signal point constellation change to predetermined times.

14. The communication method of claim **13**, wherein said step of monitoring said energy of said first signal comprises monitoring an amplitude of said first signal.

15. The communication method of claim **14**, wherein said step of monitoring comprises taking an average of said amplitude over a predefined period of time and comparing said average with a threshold.

16. The communication method of claim **13**, wherein said step of monitoring said energy comprises monitor-

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ing a peak amplitude of said first signal over a predefined period of time.

17. The communication method of claim 13, wherein said step of monitoring said energy of said first signal comprises comparing an amplitude of said first signal with a threshold.

18. The communication method of claim 13, wherein said step of monitoring said energy comprises monitoring an orthogonal component of said first signal.

19. The communication method of claim 18, wherein said step of monitoring said energy of said first signal comprises monitoring an amplitude of said orthogonal component.

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20. The communication method of claim 19, wherein said step of monitoring comprises taking an average of said amplitude over a predefined period of time and comparing said average with a threshold.

21. The communication method of claim 18, wherein said step of monitoring said energy comprises monitoring a peak amplitude of said orthogonal component over a predefined period of time.

22. The communication method of claim 18, wherein said step of monitoring said energy of said first signal comprises comparing an amplitude of said orthogonal component with a threshold.

* * * * *

EXHIBIT B

US005475691A

United States Patent [19]

Chapman et al.

[11]   Patent Number: **5,475,691**

[45]   **Date of Patent:**   **Dec. 12, 1995**

[54]   **VOICE ACTIVATED DATE RATE CHANGE IN SIMULTANEOUS VOICE AND DATA TRANSMISSION**

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[52]   U.S. Cl. 370/110.4; 375/216; 375/264

[58]   Field of Search 370/20, 69.1, 76, 370/84, 85.7, 95.1, 110.1, 110.4, 111, 125; 379/97, 98; 375/5, 39, 216, 261, 264; 455/54.2, 58.1, 58.2, 82

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Attorney, Agent, or Firm—Joseph J. Opalach

[57]   **ABSTRACT**

A simultaneous voice and data modem performs "voice-activated" data rate changes to improve the transmission quality of the voice signal. In particular, when the simultaneous voice and data modem detects that a local telephone set has gone "off-hook," the simultaneous voice and data modem selects a signal space with a lower symbol density, which, although resulting in fewer bits per symbol, provides for a higher quality voice transmission.

11 Claims, 6 Drawing Sheets
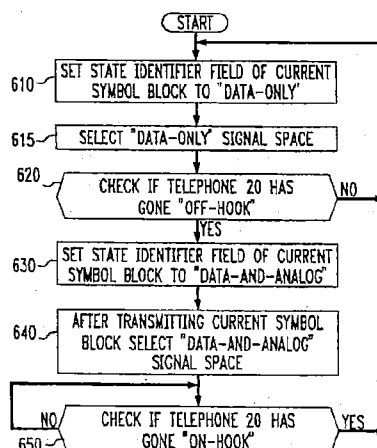
FIG. 1
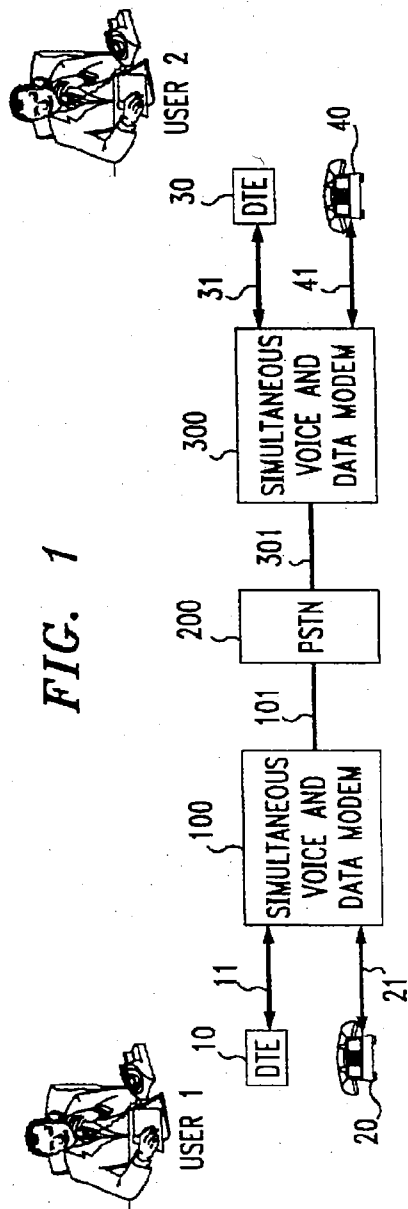
FIG. 2
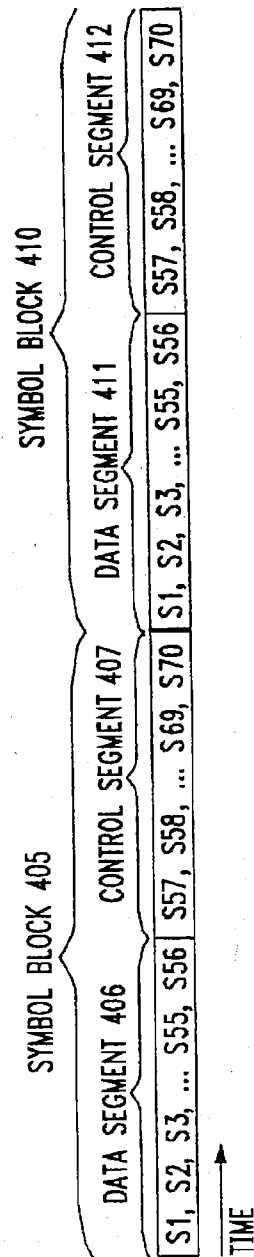
FIG. 3

CONTROL BIT ASSIGNMENTS, 3000S/SEC
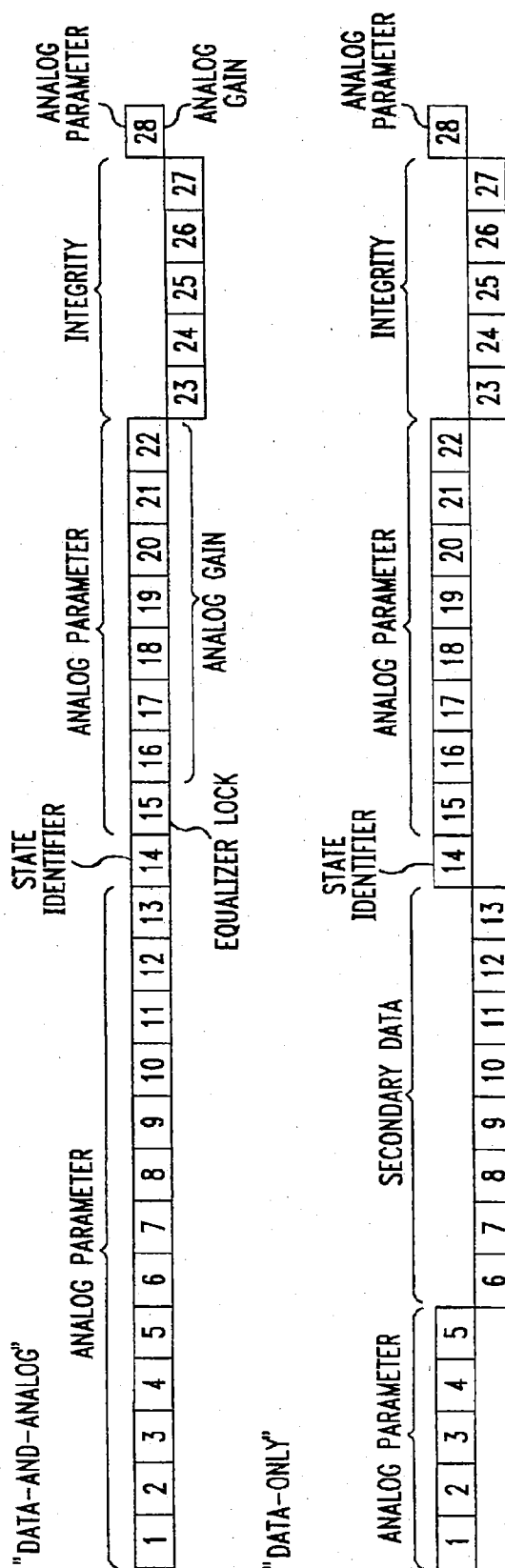
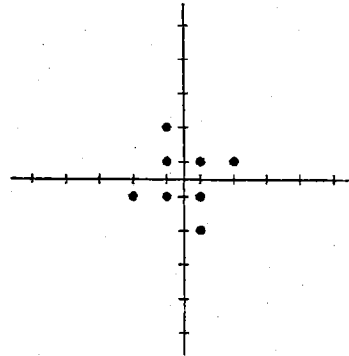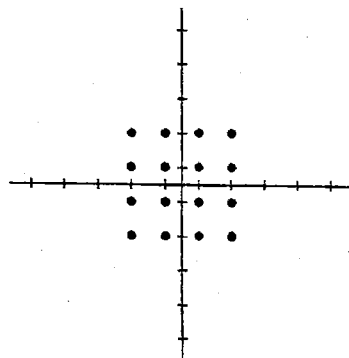# FIG. 4

CONSTELLATION A
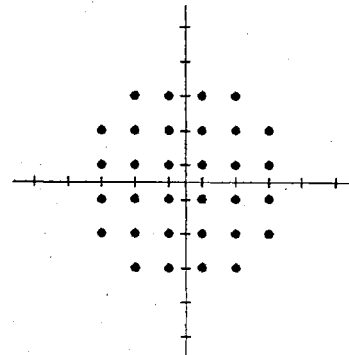TWO BITS PER SYMBOL
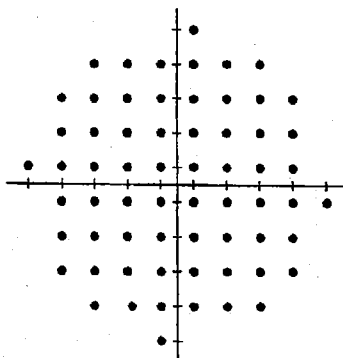4800 bps
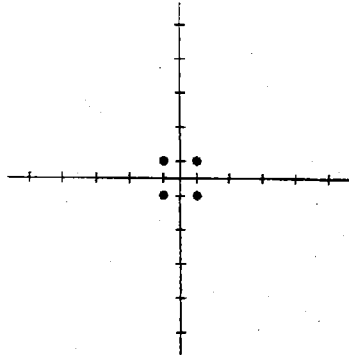
CONSTELLATION B
THREE BITS PER SYMBOL
7200 bps

CONSTELLATION C
FOUR BITS PER SYMBOL
9600 bps

CONSTELLATION D
FIVE BITS PER SYMBOL
12000 bps

CONSTELLATION E
SIX BITS PER SYMBOL
14400 bps
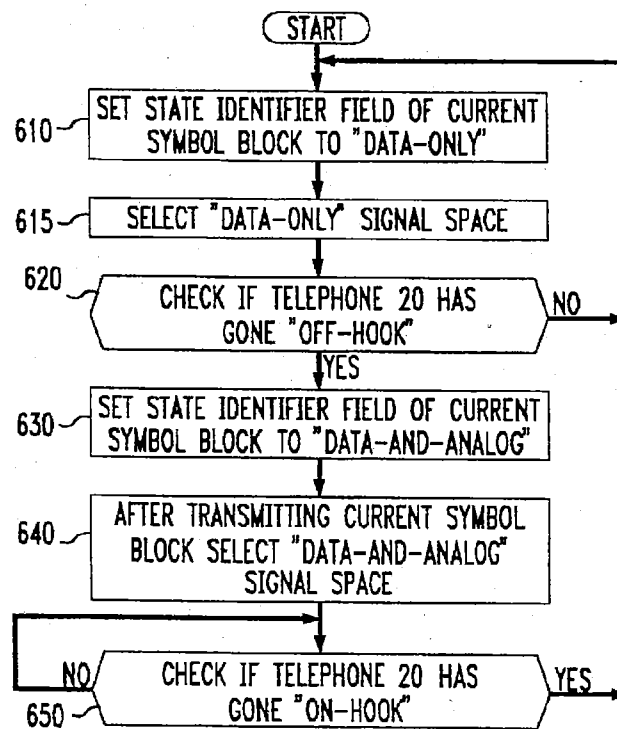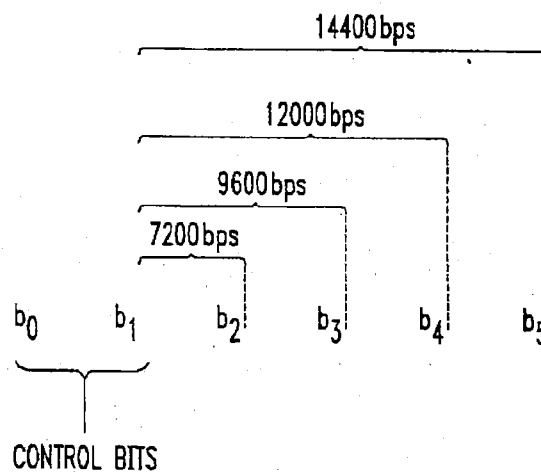
FIG. 5

START

610 — SET STATE IDENTIFIER FIELD OF CURRENT SYMBOL BLOCK TO "DATA-ONLY"

615 — SELECT "DATA-ONLY" SIGNAL SPACE

620 — CHECK IF TELEPHONE 20 HAS GONE "OFF-HOOK"   NO

YES

630 — SET STATE IDENTIFIER FIELD OF CURRENT SYMBOL BLOCK TO "DATA-AND-ANALOG"

640 — AFTER TRANSMITTING CURRENT SYMBOL BLOCK SELECT "DATA-AND-ANALOG" SIGNAL SPACE

NO   CHECK IF TELEPHONE 20 HAS GONE "ON-HOOK"   YES

650

FIG. 6

14400 bps

12000 bps

9600 bps

7200 bps

b_0 b_1 b_2 b_3 b_4 b_5

CONTROL BITS

FIG. 7
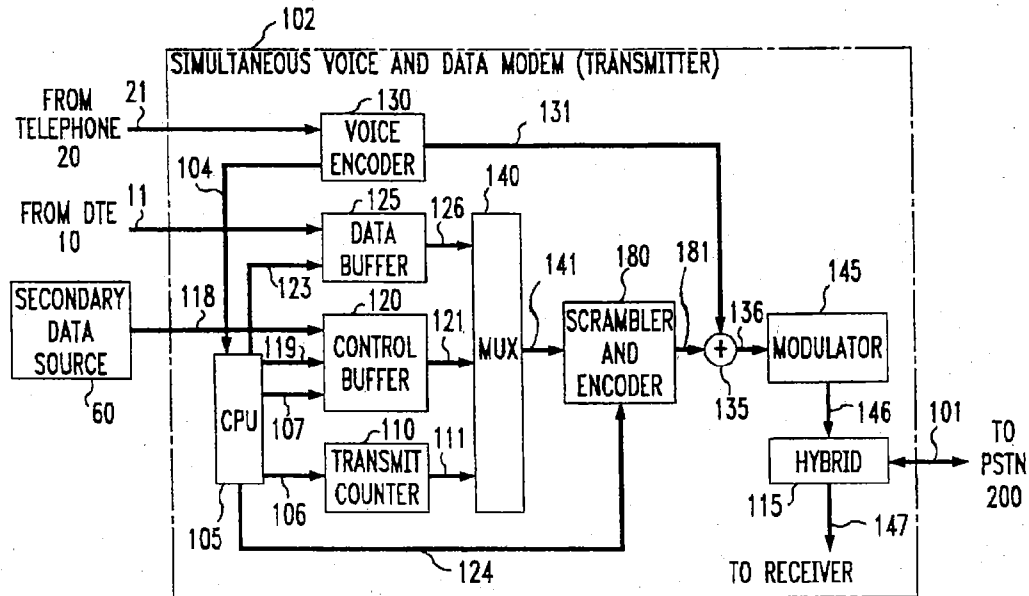
FIG. 8
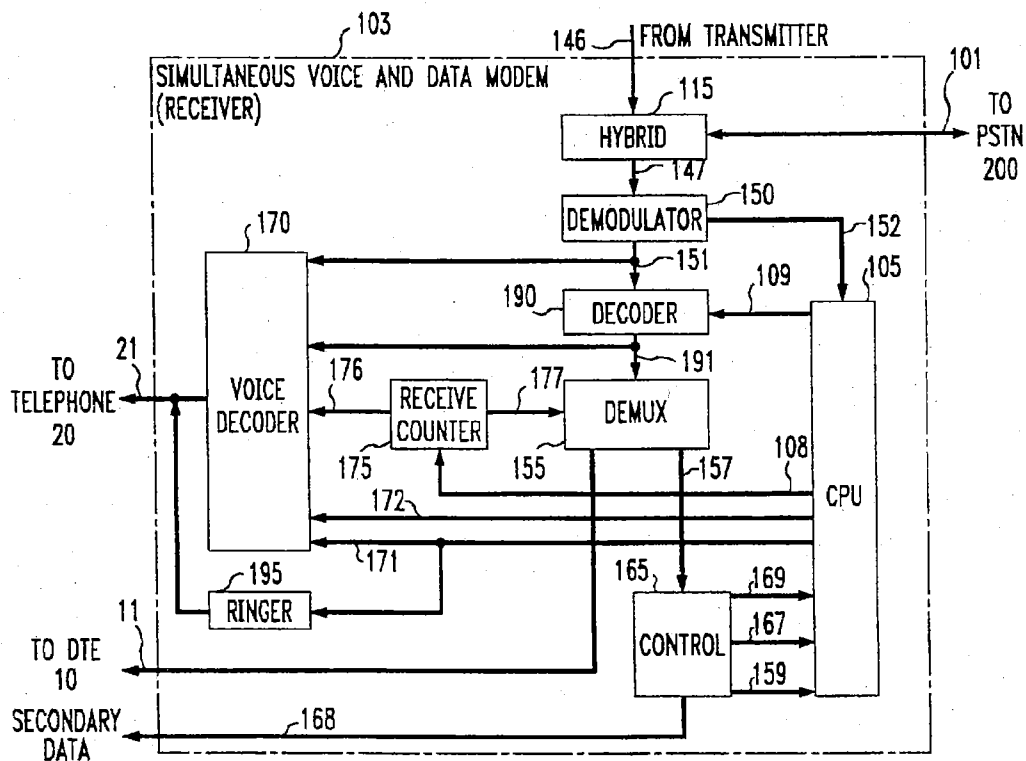
FIG. 9

FIG. 10

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VOICE ACTIVATED DATE RATE CHANGE IN SIMULTANEOUS VOICE AND DATA TRANSMISSION
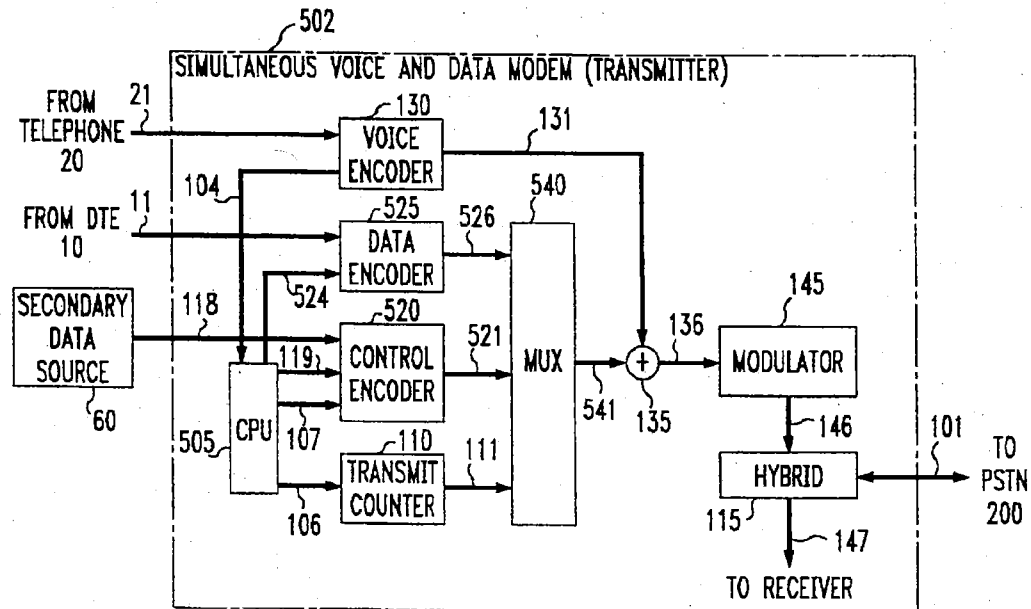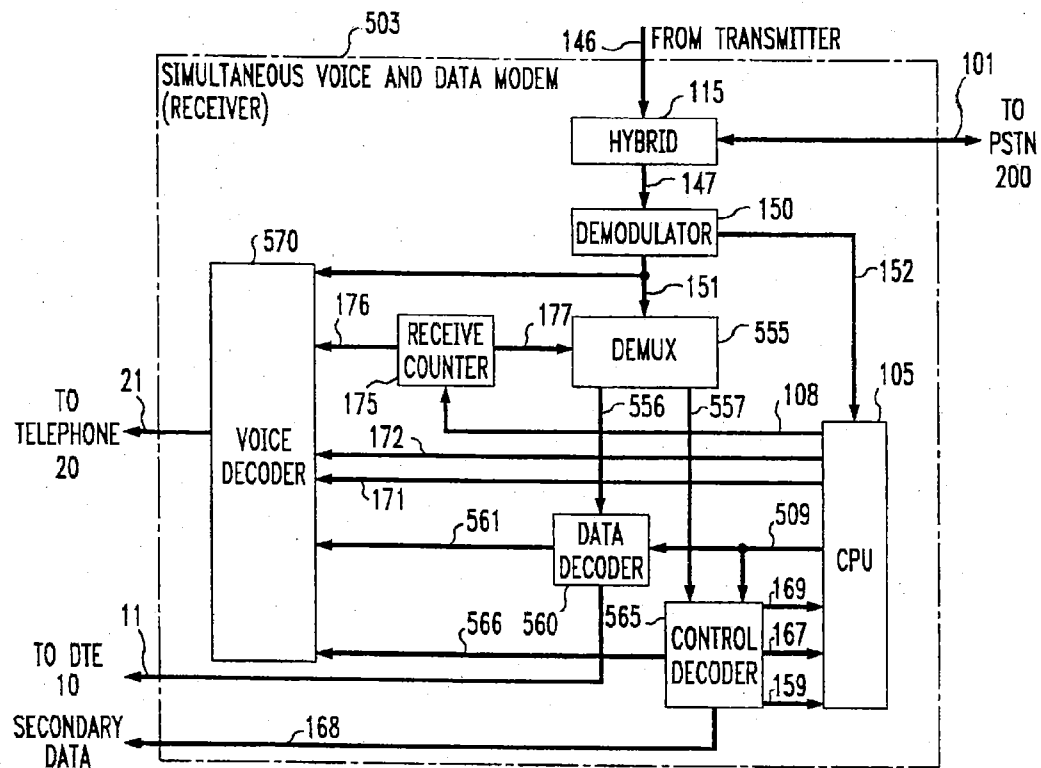
BACKGROUND OF THE INVENTION

The present invention relates to data communications equipment and, more particularly, to the detection of a voice signal in a simultaneous voice and data communications system.

The co-pending, commonly assigned, U.S. patent application of Gordon Bremer and Kenneth D. Ko entitled "Simultaneous Analog and Digital Communication," Ser. No. 08/076505, filed on Jun. 14, 1993, describes a simultaneous voice and data communications system in which a voice signal is added to a data signal for transmission over a communications channel to a receiving modem.

In this simultaneous analog and digital communication system, the data signal to be transmitted is represented by a sequence of data symbols, where each data symbol is associated with a particular N-dimensional signal point value taken from a signal space. Similarly, the analog signal, which is represented by a voice signal, is processed so that it is mapped into the N-dimensional signal space to provide a voice signal point. This voice signal point defines the magnitude and angle of a voice signal vector about the origin of the signal space. The data symbol and the voice signal vector are then added together to select a resultant N-dimensional signal point, which is then transmitted to a far-end modem.

Upon reception of the transmitted N-dimensional signal point, the receiver of the far-end modem detects the embedded data symbol and subtracts the data symbol from the received N-dimensional signal point to yield the voice signal vector. This voice signal vector is then used to recreate the voice signal.

Using this above-mentioned technique to simultaneously transit voice and data, the quality of the voice signal is affected by the underlying data rate as described in the co-pending, commonly assigned, U.S. patent application of Gordon Bremer, Kenneth D. Ko, and Luke J. Smithwick, entitled "Shaped Signal Spaces in a Simultaneous Voice and Data System," Ser. No. 08/076530, filed on Jun. 14, 1993. Generally speaking, when transmitting data plus voice the higher the data rate—the lower the quality of voice transmission. This is because at a higher transmission rate, the data symbols are closer together, which results in a lower dynamic range for the voice signal. Therefore, when transmitting a voice signal, it is preferable to reduce the number of symbols in the data signal space in order to increase the allowable dynamic range of the voice signal—which concomitantly improves the quality of the voice transmission.

SUMMARY OF THE INVENTION

A simultaneous voice and data modem performs "voice-activated" data rate changes as a function of the presence of the voice, i.e., analog, signal. This improves the transmission quality of the voice signal by changing the signal space to one that allows for a greater dynamic range of the voice signal.

In an embodiment of the invention, when a simultaneous voice and data modem detects that a local telephone set has gone "off-hook," the simultaneous voice and data modem selects a signal space with a lower symbol density, which, although resulting in fewer bits per symbol, provides for a

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higher quality voice transmission. As a result, the simultaneous voice and data modem switches from a "data-only" state to a "data-and-analog" state. The simultaneous voice and data modem transmits the state information via a side channel to a compatible far-end modem.

BRIEF DESCRIPTION OF THE DRAWING

FIG. 1 shows a block diagram of a simultaneous voice and data communications system embodying the principles of the invention;

FIG. 2 shows an illustrative symbol block;

FIG. 3 is an illustration of the control bit assignments for a control segment of a symbol block for both the "data-only" and "data-and-analog" states;

FIG. 4 shows a number of illustrative signal spaces with different bit rates;

FIG. 5 is an illustrative flow diagram of a "voice activated" data rate change method embodying the principles of the invention;

FIG. 6 is an illustration of the availability of redundant bits for the control segment at data rates higher than 4800 bps;

FIG. 7 shows a block diagram of the transmitter portion of a simultaneous voice and data modem embodying the principles of the invention;

FIG. 8 shows a block diagram of the receiver portion of a simultaneous voice and data modem embodying the principles of the invention;

FIG. 9 shows a block diagram of another embodiment of the transmitter portion of a simultaneous voice and data modem; and

FIG. 10 shows a block diagram of another embodiment of the receiver portion of a simultaneous voice and data modem.

DETAILED DESCRIPTION

A block diagram of a simultaneous voice and data communications system is shown in FIG. 1. In the description that follows it is assumed that a communications path has already been established between user 1 and user 2 of FIG. 1. The communications equipment of user 1 includes data terminal equipment (DTE) 10, telephone 20, which represents voice equipment, and simultaneous voice and data (SVD) modem 100. The latter receives two types of signals for transmission to SVD modem 300—a data signal from DTE 10 and a voice signal from telephone 20. SVD modem 100 encodes both the data signal and the voice signal to provide a combined voice and data signal for transmission, via local loop 101, public switched telephone network (PSTN) 200, and local loop 301, to SVD modem 300. The basic operation of a simultaneous voice and data modem, other than the inventive concept, is described in the above-mentioned Bremer et al. patent application entitled "Simultaneous Analog and Digital Communication," Ser. No. 08/076505, filed on Jun. 14, 1993, which is hereby incorporated by reference. SVD modem 300 receives the combined voice and data signal transmitted by SVD modem 100 and provides the data signal to DTE 30, and the voice signal to telephone 40. Transmission of data and voice signals in the opposite direction, i.e., from SVD modem 300 to SVD modem 100, occur in a like fashion. In the description that follows only SVD modem 100 is described, however, it is assumed that SVD modem 300 also incorporates the inventive concept.

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FIG. 2 shows a diagram of a transmission scheme that includes a side-channel within an SVD signal. This SVD side-channel not only provides for the transport of additional information between the SVD endpoints of FIG. 1—but also allows the voice signal to be transmitted across the full bandwidth of the SVD data connection. As can be observed from FIG. 2, information from an SVD modem is provided in a frame, or "symbol block," e.g., symbol block **405**. For the purposes of this example, a symbol block comprises **70** symbols. Consecutive symbols within each symbol block are identified as S1, S2, S3, . . . , S70.

Each symbol block is further divided into a data segment, e.g., data segment 406; and a control segment, e.g., control segment 407. Let the group of symbols in the data segment be, for example, S1 to S56. These are the "data symbols" and always convey DTE data. For the purposes of the following discussion the symbol rate is illustratively 3000 symbols/second (s/sec.), although other symbol rates may be used, e.g., 2800 s/sec. At a symbol rate of 3000 s/sec., the average data symbol rate of a symbol block is equal to $(56/70)\times 3000)=2400$ s/sec. Consequently, if there are 6 bits of data per data symbol, the resultant data rate is 14400 bits/sec (bps). It is assumed that this data rate is high enough to meet a user's needs so that the remaining bandwidth of the SVD data connection can be allocated to the control segment, which provides the side-channel.

The remaining symbols of the control segment, i.e., S57 to S70, are the "control symbols." Usually, the latter never convey DTE data, but convey control information. Each control symbol represents a number of "control bits." The control symbols are encoded and scrambled the same as the DTE data symbols, e.g., they use the same signal space. The control symbols provide the side-channel for conveying additional information between SVD modem **100** and SVD modem **300**. Although the data symbols represent user data and the control symbols represent control information, both the data and control symbols may also convey analog data, which in this example is the voice signal that is provided to SVD modem **100** by telephone **20** (described below). As a result, the side-channel is a part of the simultaneous voice and data transmission.

It should be noted that if a lower symbol rate is used, e.g., 2800 s/sec., the size of the data segment and control segment changes. For example, if it is assumed that both the size of each symbol block is fixed at 70 symbols and that the average symbol rate of 2400 s/sec. should be maintained, then a symbol block at 2800 s/sec. has a data segment of 60 symbols and a control segment of 10 symbols.

Although the symbols of the control segment can represent any type of information, in this illustrative example the control information is further divided as shown in FIG. 3 to represent analog parameter information, a state identifier, secondary data, and an integrity field. The number of bits available to represent the analog parameter information is a function of both the "state" of the symbol block and the number of bits per control symbol (described below).

The "state" of a symbol block is represented by the value of the "state identifier field" of the previous symbol block. For example, the state of symbol block **410** of FIG. 2 is defined by the value of the state identifier field of symbol block **405**, which preceded symbol block **410**. The state of any symbol block in this example is limited to two: "data-only" or "data-and-analog." Consequently, the state identifier field is conveniently represented by one bit, illustratively control bit number **14** shown in FIG. 3. A value of "one" represents the "data-and-analog" state, while a value of

"zero" represents the "data-only" state.

SVD modem **100** defaults to the "data-only" state, e.g., upon power-up, and the state identifier bit is initially set to represent the "data-only" state in any subsequent data connection established with SVD modem **300**. FIG. 4 shows a number of illustrative signal spaces for transmitting information between the SVD modem endpoints. Although the data symbols represent user data and the control symbols represent control information, both the data and control symbols are selected from the same signal space. In the "data-only" state any of these 6 signal spaces can be used to transmit information between the SVD endpoints. Constellation "A" of FIG. 4 shows a signal space where each symbol represents two bits of information. Similarly, constellation "E" illustrates a signal space where each symbol represents 6 bits of information. Preferably, constellation E of FIG. 4 will be used since, conditions permitting, it allows the highest transmission bit rate between the SVD endpoints.

There is voice transmission between SVD endpoints only in the "data-and-analog" state. As noted above, when simultaneously transmitting both voice and data, there is a tradeoff between the quality of the voice transmission and the size of the symbol constellation. For example, if constellation E of FIG. 4 is used in the "data-and-analog" state, the higher symbol density reduces the dynamic range of any voice signal that is superimposed on any of the symbols taken from this constellation—with the result that voice quality is impaired. Therefore, it is desirable to select constellation A during voice transmission because the fewer symbols of constellation A allow a larger dynamic range for the voice signal—and therefore improves the quality of the voice signal transmission.

In accordance with the inventive concept, an illustrative method for use in SVD modem **100** for "voice activated" data rate changes is shown in FIG. **5**. At power-up, or the beginning of each data connection, SVD modem **100** enters the "data-only" state in step **610**. In step **615**, SVD modem **100** selects that signal space associated with the "data-only" state, i.e., the highest data rate negotiated between SVD modem **100** and SVD modem **300**. SVD modem **100** monitors telephone **20** in step **620** to check if telephone **20** has gone "off-hook." As long as user **1** does not go off-hook, SVD modem **100** remains in the "data-only" state by returning to step **610**. However, when SVD modem **100** detects that user **1** has taken telephone **20** off-hook, SVD modem **100** assumes that voice communications is desired and switches to the "data-and-analog" state. The use of a side-channel provides the ability to relay this information as to the current data signal space to far-end SVD modem **300**. In step **630**, SVD modem **100** alters the state identifier bit in the current symbol block, e.g., block **405** of FIG. **2**, to indicate to SVD modem **300** that the next symbol block, e.g., block **410**, will be in the "data-and-analog" state. Upon completion of transmitting symbol block **405**, SVD modem **100** then switches to constellation A of FIG. **4** in step **640** for the transmission of symbol block **410**. Thus, when user **1** goes off-hook at telephone **20**, SVD modem **100** dynamically changes the bit rate to accommodate the presence of a voice signal. Consequently, upon reception of block **405**, SVD modem **300** not only knows which signal space to use to decode the incoming symbol stream from block **410**, but also can infer the state of the switch hook at telephone **20**.

After switching to the "data-and-analog" state, SVD modem **100** monitors telephone **20** to detect that user **1** has gone "on-hook" in step **650**. When user **1** goes "on-hook," SVD modem **100** returns to step **610** to set the state identifier

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field for the next symbol block to the "data-only" state and thereafter switches data rates back to the last data rate negotiated between SVD modem **100** and SVD modem **300**.

Returning to FIG. **3**, the number of control bits in the control segment is shown as fixed at 28 bits. However, generally speaking, assuming a constant symbol rate, the number of control bits available for the control segment varies with the number of bits per control symbol. For example, in the "data-only" state any of the constellations of FIG. **4** may be used. If constellation E is used, there are 6 bits per symbol. Therefore, there are 84 bits available in any control segment for transporting control information between SVD modem **100** and SVD modem **300**. However, in the "data-and-analog" state, SVD modem **100** switches to constellation A, which results in only 2 bits per symbol, or 28 bits in the control segment for the 14 control symbols S57 to S70. Consequently, while the number of bits in a control segment could indeed vary as a function of the selected signal space, in this embodiment, the number of bits in the control segment is bounded by the number of control bits available in the "data-and-analog" state, i.e., 28 bits.

In the "data-and-analog" state, control bits N1 to N13, N15 to N22, and N28, are used to represent "analog parameter" information. Similar to the state indicator field, these analog parameters convey information pertaining to the next symbol block like "adaptive gain" information in bit locations **15–22** and **28**. In this example, the analog parameter bit locations **1–13** in the "data-and-analog" state are reserved for future use. It should be noted that information conveyed by the control segment does not have to be restricted to information about the "next" symbol block.

In a "data-only" symbol block, the transmission of "analog parameters" is optional. In fact, the number of control bits available for the transport of analog parameter information is reduced so that in the "data-only" state secondary data is transmitted by control bits N6 to N13. Any transmission of secondary data does not use all of the control bit assignments even though there is no "analog parameter" information transmitted in a "data-only" state. Using these control bit assignments, the secondary data rates at 3000 s/sec. is 342 bits per sec.

As noted above, in the "data-only" state higher density signal spaces may be used with the result that there are more bits available for transport of control information than are actually used in the above-defined control segment. However, there is nevertheless a way to use these additional data bits in the SVD communications system of FIG. **1** to minimize errors in detecting the correct state of a received SVD symbol block.

If an SVD receiver makes an error in detecting the correct state of the received symbol block, this error has different effects on a user depending on the correct state. For example, if the state of a received symbol block is "data-only" but the receiver interprets the state as "data-and-analog," then the receiver will decode data with perhaps the wrong decision regions and will enable the analog output—causing a "data blast" to the listener. Conversely, if the state is "data-and-analog" but the receiver interprets the state as "data-only" then the receiver will inadvertently silence the analog output. Of these two possible error conditions, it is likely that an erroneous decision that the symbol block is in the "data-only" state is more acceptable to a user since the user will only hear silence. However, a mistaken decision that the symbol block is in the "data-and-analog" state may be more annoying to a user because of the likelihood a user will hear a data blast. Therefore, it would be better if the possibility

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of this latter type of error were minimized. Especially since the possibility of this type of error increases as the data rate increases.

To lessen the possibility of an erroneous detection of the "data-and-analog" state while in the "data-only" state, some of the additional bits heretofore unused in each control symbol, at data rates higher than 4800 bps, are now utilized as "redundant" state identifier bits. In particular, one additional bit per control symbol is used at data rates greater than 4800 bps. This provides an additional fourteen bits of information, albeit redundant, to an SVD receiver. FIG. **6** shows a simple illustration of this technique. While in the "data-only" state, data transmission can occur at data rates from 4800 bps to 14400 bps as provided by the signal spaces shown in FIG. **4**. Each control symbol therefore represents a number of bits that is a function of the currently selected signal space. This number of bits varies from b_0 to b_5 as shown in FIG. **6**. The data rate of 4800 bps provides the minimum number of bits per symbol—b_0 and b_1. As the data rate increases, additional bits become available, e.g., one bit, b_2, at a data rate of 7200 bps, and upto four bits, b_2 to b_5, at a data rate of 14400 bps. In this illustrative embodiment, only one additional bit, e.g., b_2, is used; any additional bit capacity at the higher data rates is simply ignored.

These fourteen bits are simply copies of the state identifier bit transmitted in the control segment of the current symbol block. The SVD receiver performs a "majority" vote of the fourteen redundant bits and the state identifier bit from the control segment to determine the appropriate state for the next symbol block.

Although at data rates higher than 4800 bps an additional redundant bit is used from the control symbols to protect against an erroneous decision by an SVD receiver as to the state of the next symbol block, at the lower data rate of 4800 bps it is assumed that the integrity field of the control segment, combined with the spatial separation provided in the 4800 bps constellation, provides adequate protection of the state identifier bit.

From FIG. **3**, it can be seen that five control bits define the integrity field. In this example, the integrity field represents the inverse of five other predefined control bits. These integrity bits are used to bolster the decoding of the control segment by an SVD receiver. The five control bits 14 through **18** are protected by the integrity field. These are the state identifier bit, the equalizer lock bit for the upcoming symbol block, and 3 bits that represent the most significant bits of adaptive gain. The SVD receiver uses biased voting to evaluate the integrity bits. For example, if received adaptive gain bit is inconsistent with its respective integrity bit, the receiver uses the gain value that results in a lower volume to the audio speaker, on the premise that a brief reduction in volume is more acceptable to the listener than an unexpected increase in volume.

As shown in FIG. **3**, the integrity field is located within the control segment, as opposed to being located at either boundary of the control segment, to increase the probability that a shift in timing will cause errors in the integrity field. An SVD receiver also keeps track of integrity errors over multiple symbol blocks as an indicator of gross channel conditions or loss of symbol counter synchronization (described below). Either of these conditions causes the SVD receiver to perform a retraining with the opposite SVD endpoint.

As described above, it is advantageous to protect against the possible occurrence of an erroneous switch from one state to the other. In the analog parameter field there is gain

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information for the voice signal in the "data-and-analog" state. However, it is not necessary to provide any gain information in the "data-only" state since there is no voice signal. Therefore, additional protection can be provided against an erroneous switch from the "data-only" state to the "data-and-analog" state by providing dummy gain information such that if an SVD receiver erroneously switches to the "data-and-analog" state the resulting amplification of the data signal, which appears as noise to a user, is low.

Reference should now be made to FIG. 7, which shows a block diagram of transmitter 102 of SVD modem 100 that embodies the principles of this invention. Other than the inventive concept, the individual components of SVD modem 100 are well-known and are not described in detail. For example, CPU 105 is a microprocessor-based central processing unit and associated memory for storing program data. Also, it is assumed that the operating data symbol rate and the number of data bits per symbol in the "data-only" state are determined during the initial training and rate negotiation sequences, and by any retraining sequences that occur between SVD modems 100 and 300. Although it is assumed that the symbol rate does not change during a communications session, the number of data bits per symbol may also change in accordance with any of the well-known autorating techniques.

Telephone 20 provides a voice signal to voice encoder 130. The latter provides a sequence of two-dimensional signal points, at a predefined symbol rate of 1/T symbols per sec., on line 131. Each two-dimensional signal point represents a "voice signal vector" about the origin of a signal space (not shown). In addition, line 104 conveys signaling to provide CPU 105 with information on the "off-hook" status of telephone 20, and analog gain information for transmission to far-end SVD modem 300. As discussed above, the "off-hook" signal alerts CPU 105 when user 1 at telephone 20 goes off-hook or on-hook so that SVD modem 100 can select between the "data-only" state or the "data-and-analog" state as described above. CPU 105 controls the selection of the appropriate signal space in scrambler and encoder 180 via the signal on line 124.

DTE 10 provides a data signal to data buffer 125, which stores the data provided by DTE 10 for latter delivery to multiplexer (MUX) 140. Control buffer 120 receives two signals on lines 107 and 118. The signal on line 118 represents any secondary data source. In fact, secondary data source 60 represents the ability of SVD modem 100 to provide additional bandwidth for data communications albeit within the control segment during the "data-only" mode of operation. Although shown for simplicity as an independent source, secondary data source 60 can be DTE 10. For example, control buffer 120 could be coupled to data buffer 125 to allocate this additional data bandwidth to DTE 10. In comparison, the signal on line 107 represents the analog parameter information as defined in FIG. 3. It can be observed that while this embodiment sends a variety of different types of control information, there is no requirement that any information peculiar to the SVD communications system be transmitted in the control segment.

The state of SVD modem 100 is provided from CPU 105 to control buffer 120 via line 119. This represents the value used by control buffer 120 for the state indicator bit of the control segment. If SVD modem 100 is in the "data-only" state, then control buffer 120 multiplexes any analog parameter information, if any, along with the secondary data to provide the control segment as shown in FIG. 3 for the "data-only" state. On the other hand, if SVD modem 100 is in the "data-and-analog" state, then control encoder 120

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provides the "data-and-analog" control segment, which only includes analog parameter information as provided by CPU 105 via line 107. Control encoder 120 also generates the integrity field and, for signal spaces which accommodate it, duplicates the state identifier bit to provide the above-described redundancy when the data rate is greater than 4800 bps.

Transmit counter 110 controls MUX 140, which provides either data or control information to scrambler and encoder 180. The latter includes any of the well-known encoding techniques like scrambling, trellis-coding, etc., to provide the sequence of symbols on line 181 at a symbol rate, 1/T. The symbols are selected from one of the signal spaces shown in FIG. 4. The selection of the signal space is controlled by CPU 105 via line 124.

CPU 105 synchronizes transmit counter 110 in response to a training, or retraining, event between SVD modems 100 and 300. As is known in the art, both modems of a data connection typically perform a hand-shaking procedure that includes a training sequence to initialize the equalizers and echo cancelers of each modem (not shown). The need to perform a retraining sequence is detected by the loss of synchronization. Loss of synchronization is detected be CPU 110 when the receiver of SVD modem 100 (discussed below) indicates that too many integrity field errors have occurred over a number of received symbol blocks.

Transmit counter 110 must be synchronized because transmit counter 110 frames the symbol blocks by counting symbol periods. As described above, and shown in FIG. 2, each symbol block includes 70 symbols. Therefore, transmit counter 110 counts "modulo 70." During the first 56 symbol periods, i.e., the data segment, transmit counter 110 controls MUX 140 to provide the data segment information to scrambler and encoder 180. During the last 14 symbol periods, i.e., the control segment, transmit counter 110 controls MUX 140 to provide the control segment information to scrambler and encoder 180.

Adder 135 adds each voice signal vector on line 131, if any, to a respective one of the symbols provided by scrambler and encoder 180 to provide a stream of signal points to modulator 145. The latter functions in accordance with the well-known quadrature amplitude modulation (QAM) to provide a transmit signal to hybrid 146 for transmission to SVD modem 300 via PSTN 200.

Receiver 103 of SVD modem 100 performs complementary functions to transmitter 102 described above and is shown in block diagram form in FIG. 8. Common elements in receiver 103 and transmitter 102 have the same reference numeral, e.g., CPU 105, hybrid 115, etc. Hybrid 115 receives a transmitted signal from SVD modem 300, via PSTN 200, and applies this received signal to demodulator 150. The latter provides a received signal point sequence to decoder 190, which performs the inverse function of scrambler and encoder 180 of transmitter 102 to provide an information-bearing signal each symbol period to demultiplexer (DEMUX) 155. Receive counter 175 controls DEMUX 155 via line 177. After a training, or retraining, event, as described above, CPU 105 resets receive counter 175 to begin counting modulo 70. Receive counter 175 controls DEMUX 155 to apply the first 56 symbol periods of information to DTE 10 via line 11. This is the data segment. Then receive counter 175 controls DEMUX 155 to apply the last 14 symbol periods of information to control element 165. Receive counter 175 repetitively continues this demultiplexing of the received information stream until reset by CPU 105.

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<table>
<tr><td>9</td><td>10</td></tr>
</table>

As noted above, each SVD modem initially begins in the "data-only" state. Consequently, receiver 103 assumes that the first symbol block received is in the "data-only" state. The state indicator field of this first symbol block then determines the state of the succeeding symbol block, etc.

Control element 165 provides CPU 105 with the value of the state indicator bit on line 167; any analog parameter information on line 169; and an indicator if there was an error in the integrity field on line 159. Control decoder 165 performs the majority vote for any duplicate state indicator bits (as described above) when the data rate is greater than 4800 bps. Additionally, control decoder 165 processes the integrity field bits (as described above). Finally, control decoder 165 provides any secondary data, via line 168.

In response to the information provided by control decoder 165, CPU 105 performs a number of actions. First, based on the value of the state indicator information, CPU 105 controls the signal space used by decoder 190 to decode the next symbol block via line 109. This allows receiver 103 to correctly demultiplex and decode the received signal point sequence. Next, CPU 105 adjusts any analog settings based upon the analog parameter information, via line 172. In this embodiment, the analog parameter information is only used by voice decoder 170. This allows receiver 103 to easily adapt gain settings for the voice signal and other analog parameters (if any). Finally, CPU 105 generates a retrain based upon the cumulative statistics of the number of errors in the integrity field over a period of time. When a retrain event occurs, CPU 105 resets receive counter 175. It should be noted that CPU 105 also resets the receive counter if a training, or retraining, sequence from the far-end SVD modem is detected by demodulator 150, which notifies CPU 105 via line 152.

Voice decoder 170 provides the voice signal during the "data-and-analog" state to telephone 20. Voice decoder 170 is enabled during the "data-and-analog" state by CPU 105 via line 171. The signal on line 171 also enables ringer 195 to alert the user of an incoming voice call. Although shown as a "ringer" element, element 195 can equivalently be any alerting mechanism to get the attention of a user that a voice call has arrived, e.g., an audible tone through a speaker-phone (not shown). Receive counter 176 provides a synchronization signal to voice decoder 170 so that the correct received symbol is subtracted from the received signal point sequence conveyed by line 151. The received symbol sequence is provided by line 191 of decoder 190. Voice decoder 170 includes buffering to accommodate any delays introduced by decoder 190 in decoding the received signal points.

The above-described inventive concept thus allows for voice-activated data rate changes that can be relayed via an SVD side-channel, where both the data and control symbols are available to carry the voice signal. Another embodiment of a transmitter and receiver is shown in FIGS. 9 and 10, respectively. FIG. 9 is a block diagram of the transmitter portion of an SVD modem and is similar to FIG. 7 except that the respective signal spaces used for the data and control segments are different.

DTE 10 provides a data signal to data encoder 525, which provides a sequence of two-dimensional signal points at the symbol rate, 1/T. These two-dimensional signal points are selected from one of the signal spaces shown in FIG. 4, which is controlled by CPU 105 via line 524. Each signal point is associated with a particular two-dimensional data symbol. Data encoder 525 includes any of the well-known encoding techniques like scrambling, trellis-coding, etc., to provide the sequence of data symbols.

Control encoder 520 receives two signals, as described above, on lines 107 and 119 and provides a sequence of control symbols on line 521 at the symbol rate, 1/T. Control encoder 520 functions in a similar fashion to data encoder 525 and includes any of the well-known encoding techniques like scrambling, trellis-coding, etc., to provide the sequence of control symbols. Similar to data encoder 525 above, these two-dimensional symbols are selected from one of the signal spaces shown in FIG. 4. These control symbols represent a control segment. The state of SVD modem 100 is provided from CPU 105 via line 119. If SVD modem 100 is in the "data-only" state, then control encoder 520 multiplexes any analog parameter information, if any, along with the secondary data to provide the control segment as shown in FIG. 3 for the "data-only" state. On the other hand, if SVD modem 100 is in the "data-and-analog" state, then control encoder 520 provides the "data-and-analog" control segment, which only includes analog parameter information as provided by CPU 105 via line 107. Control encoder 520 also generates the integrity field and, in the "data-only" state, duplicates the state identifier bit to provide the above-described redundancy when the data rate is greater than 4800 bps.

Transmit counter 110, which counts modulo 70, controls multiplexer (MUX) 540, which provides the above mentioned symbol blocks on line 541. CPU 105 synchronizes transmit counter 110 in response to a training, or retraining, event between SVD modems 100 and 300. During the first 56 symbol periods, i.e., the data segment, transmit counter 110 controls MUX 540 to provide the data symbols from data encoder 525 to adder 135. During the last 14 symbol periods, i.e., the control segment, transmit counter 110 controls MUX 540 to provide the control symbols from control encoder 520 to adder 135. Since transmit counter 110 switches MUX 540 between data encoder 525 and control encoder 520, each of these encoders must include buffers to provide storage for any data accumulated during that period of time when the other encoder is providing symbols to MUX 540.

Adder 135 adds each voice signal vector on line 131, if any, to a respective one of the symbols provided by MUX 540 to provide a stream of signal points to modulator 145. The latter functions in accordance with the well-known quadrature amplitude modulation (QAM) to provide a transmit signal to hybrid 146 for transmission to SVD modem 300 via PSTN 200.

Receiver 503, shown in FIG. 10, performs complementary functions to transmitter 502 of FIG. 9. Hybrid 115 receives a transmitted signal from SVD modem 300, via PSTN 200, and applies this received signal to demodulator 150. The latter provides a received signal point sequence to demultiplexer (DEMUX) 555, which is controlled by receive counter 175 via line 177. After a training, or retraining, event, as described above, CPU 105 resets receive counter 175 to begin counting modulo 70. Receive counter 175 controls DEMUX 555 to apply the first 56 received signal points to data decoder 560. Then receive counter 175 controls DEMUX 555 to apply the last 14 received signal points of the received symbol block to control decoder 565. Receive counter 175 repetitively continues this demultiplexing of the received signal point stream until reset by CPU 105.

Control decoder 565 provides CPU 105 with the value of the state indicator bit on line 167; any analog parameter information on line 169; and an indicator if there was an

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error in the integrity field on line **159**. Control decoder **565** performs the majority vote for any duplicate state indicator bits (as described above) when the data rate is greater than 4800 bps. Additionally, control decoder **565** processes the integrity field bits (as described above). Finally, control decoder **565** provides any secondary data, via line **168**.

In response to the information provided by control decoder **565**, CPU **105** performs a number of actions. First, based on the value of the state indicator information, CPU **105** controls the signal space used by data decoder **560** and control decoder **565** to decode the next symbol block via line **109**. This allows receiver **503** to correctly demultiplex and decode the received signal point sequence. Next, CPU **105** adjusts any analog settings based upon the analog parameter information, via line **172**. Finally, CPU **105** resets receive counter **175** in response to a training, or retraining, event as described earlier.

Both data decoder **560** and control decoder **565** perform the inverse of the coding functions of data encoder **525** and control encoder **520**, respectively. Voice decoder **570** provides the voice signal during the "data-and-analog" state to telephone **20**. Voice decoder **570** is enabled during the "data-and-analog" state by CPU **105** via line **171**. Receive counter **176** provides a synchronization signal to voice decoder **570** so that the correct received symbol is subtracted from the received signal point sequence conveyed by line **15** **1**. Voice decoder **570** includes buffering to accommodate any delays introduced by data decoder **160** and control decoder **165** in decoding the received symbols.

An SVD symbol block conveys either synchronous data, or asynchronous data, streams. However, the use of an SVD symbol block makes it possible to send "raw" asynchronous data (defined below) without sending the start and stop bits of this data. The start and stop bits are removed after a character is received from the DTE and restored at the other end of the circuit before sending the character to the DTE. The framing provided by the SVD block coding is used to preserve the character boundaries so that they can be properly reconstructed at the receiving end of the link. The ability to send asynchronous formatted data without the start and stop bits provides a significant improvement in the overall responsiveness of the system and effectively increases the available data bandwidth by 25%.

The term "raw" asynchronous data means that the modem is configured so that the modem's own error control and data compression capabilities are not used. In this mode, commonly referred to as "buffered mode," characters received from the DTE are sent bit for bit to the other modem. Because flow control mechanisms are still available, the data rate between the DTE and the modem can be different from the rate used over the PSTN line, however the data content of the two data streams are identical.

In this embodiment, this raw data mode is enabled by CPU **105** in response to DTE **10** providing a "buffered mode" command. As in known in the art, data terminal equipment, like DTE **10**, can configure or control various options in a modem, like SVD modem **100**, by putting the modem in a "command mode." During the command mode of operation, the modem interprets data from the data terminal equipment as instructions for the modem. A user of a modem may enter the command mode in a number of ways, e.g., by applying power to the modem, or by sending to the modem a predefined sequence of characters, like "+++" as defined in the "AT command set." For the purposes of this discussion, SVD modem **100** provides a command mode that is similar to the "AT command set."

Referring back to FIG. 7, after receiving the buffered modem command, CPU **105** of SVD modem **100** provides not only the current state identifier to data buffer **125**, but also uses line **123** to provide a signal to data buffer **125** to strip the start and stop bits from any data provided by DTE **10**. One bit from the analog parameter field is then used to identify this raw data mode to specify that the next data segment includes raw data. As a result, receiving SVD modem **300**, upon detecting this information in the analog parameter field, then controls DEMUX **155** to reconstruct the data bytes by adding the start and stop bits before sending the data to DTE **10**.

This data transfer configuration may be performed for use by "telegraphics" programs which are PC applications that are designed for efficient transfer of graphical information over PSTN circuits. These programs must communicate with the modem using asynchronous data formats because personal computers are not equipped with the interfacing hardware needed for more bandwidth-efficient synchronous transmission.

For an SVD symbol block in which the number of data symbols is an integral number of eight, no additional framing information is needed since, regardless of the number of data bits per symbol, each block contains an integral number of octets derived from asynchronous characters by stripping the start and stop bits. However, for an SVD symbol block in which the number of symbols is not an integral multiple of eight, a "super-frame" structure is required. This requires reserving at least one bit of the available "analog parameter bits" (or some unique pattern of these bits) to periodically mark the beginning of the super-frame.

As described above, a voice-activated data rate change is signaled by the state of the local switch-hook. This state information is subsequently communicated to a far-end SVD modem via the state indicator bit of the command segment. An alternative approach to the use of a side-channel would be to communicate this status information as part of a training, or retraining, event. In particular, when a local SVD modem detected an off-hook signal from the local telephone, the local SVD modem would then signal a retrain to the far-end SVD modem. The retrain signal would include a predefined bit pattern to signal the far-end modem that, unlike a normal retraining event, this retrain is only to switch data rates because of the presence of a voice signal.

The foregoing merely illustrates the principles of the invention and it will thus be appreciated that those skilled in the art will be able to devise numerous alternative arrangements which, although not explicitly described herein, embody the principles of the invention and are within its spirit and scope.

For example, although the invention is illustrated herein as being implemented with discrete functional building blocks, e.g., encoders, decoders, transmitter, etc., the functions of any one or more of those building blocks can be carried out using one or more appropriate programmed processors, e.g., a digital signal processor.

In addition, the analog signal is not limited to a voice signal, any analog signal can be used, e.g., even another data-bearing signal. The order of the control and data segments does not matter, and the inventive concept is applicable to any N-dimensional signal space. Other techniques can be used to select the appropriate signal space as a function of the state of an SVD modem, e.g., detecting the presence of voice energy from the telephone. Also, the integrity field is not limited to an inverse bit technique, other error detection schemes, like parity, can be performed on a

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portion, or all, of the control segment. Finally, while FIG. 1 illustratively coupled simultaneous voice and data modem **100** to DTE **10** and telephone **20** via local loop **101**, the inventive concept is also applicable to other communications environments like cellular.

We claim:

1. A method for use in data communications equipment, the method comprising the steps of:

selecting a signal space as a function of the detection of an analog signal;

selecting symbols from the selected signal space as a function of an applied data signal;

providing a resultant signal, where the resultant signal represents the selected symbols when no analog signal is detected and represents the addition of a respective signal point to each one of the selected symbols when the analog signal is detected; and

transmitting the resultant signal.

2. The method of claim **1** wherein the analog signal is a voice signal.

3. The method of claim **2** wherein the detection of the voice signal is provided by an off-hook signal.

4. A method for changing a data rate comprising the steps of:

selecting a first signal space when no off-hook signal is detected;

transmitting a data signal at a first data rate, where the data signal is represented by a stream of data symbols selected from the first signal space, the data symbols having k bits per symbol;

selecting a second signal space when an off-hook signal is detected wherein the data symbols have j bits per symbol and j<k, wherein the data signal is now represented by a stream of data symbols selected from the second signal space;

encoding a voice signal to provide a stream of signal points subsequent to the detection of the off-hook signal; and

adding the stream of signal points to the stream of data symbols selected from the second signal space to provide a stream of data plus voice signal points;

wherein the step of transmitting now transmits the data plus voice signal points.

5. Modem apparatus comprising:

means for selecting a signal space as a function of the presence of an analog signal;

means for selecting symbols from the selected signal space as a function of an applied data signal;

means for adding to each one of the selected symbols a respective signal point that represents the analog signal if the analog signal is present to provide a sequence of resultant signal points; and

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means for transmitting the resultant sequence of signal points.

6. The apparatus of claim **5** wherein the analog signal is a voice signal.

7. The apparatus of claim **6** wherein the means for selecting a signal space operates in response to detecting an off-hook signal, which represents the presence of the voice signal.

8. Apparatus for use in a simultaneous voice and data modem, the apparatus comprising:

means for coupling the modem to a data signal source, an analog signal source and a communications channel;

means responsive to the detection of an analog signal from the analog signal source for providing a resultant signal, wherein if no analog signal is detected, the resultant signal represents a sequence of data symbols selected from a first signal space, and if the analog signal is detected the resultant signal represents a sequence of signal points, where each signal point has a data symbol component and an analog signal component, where the data symbol component is selected from a second signal space; and

means for transmitting the resultant signal to a far-end modem over the communications channel.

9. The apparatus of claim **8** wherein the analog signal is a voice signal.

10. The apparatus of claim **9** wherein the analog signal source is a telephone and further including a means of detecting the presence of the analog signal by the off-hook status of the telephone.

11. Data communication equipment apparatus:

means responsive to an off-hook signal for selecting a first signal space when an off-hook signal is not present and for selecting a second signal space when the off-hook signal is present, where data symbols of the first signal space have k bits per symbol and data symbols of the second symbol space have j bits per symbol and j<k;

means for encoding a voice signal after the off-hook signal is detected to provide a stream of voice signal points;

means for encoding a data signal to provide a stream of data symbols from the selected signal space;

means for adding the stream of signal points to the stream of data symbols after the off-hook signal is detected to provide a stream of data plus voice signal points; and

means for transmitting the data signal as represented by either the stream of data symbols when the off-hook signal is not present or the stream of data plus voice signal points after the off-hook signal is detected.

*   *   *   *   *

UNITED STATES PATENT AND TRADEMARK OFFICE
CERTIFICATE OF CORRECTION

PATENT NO.   :   5,475,691
DATED   :   December 12, 1995
INVENTOR(S) :   Chapman et al.

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

In the Title delete "DATE" and replace with --DATA--.
Column 1, line 39, delete "transit" and replace with --transmit--.
Column 5, line 40, delete "rates" and replace with --rate--.
Column 6, line 21, delete "upto" and replace with --up to--.
Column 7, line 67, delete "encoder" and replace with --buffer--.
Column 8, line 3, delete "encoder" and replace with --buffer--.
Column 8, line 23, delete "be" and replace with --by--.
Column 8, line 24, delete "110" and replace with --105--.
Column 8, line 44, delete "146" and replace with --115--.
Column 9, line 42, delete "176" and replace with --175--.
Column 10, line 47, delete "146" and replace with --115--.
Column 11, line 24, delete "176" and replace with --175--.
Column 11, line 28, delete "160" and replace with --560--.
Column 11, line 29, delete "165" and replace with --565--.

Signed and Sealed this

Twenty-eighth Day of January, 1997

Attest:

BRUCE LEHMAN

Attesting Officer   Commissioner of Patents and Trademarks

EXHIBIT C

US005475713A

United States Patent [19]

Bremer et al.

[11] Patent Number: 5,475,713

[45] Date of Patent: Dec. 12, 1995

[54] **SHAPED SIGNAL SPACES IN A SIMULTANEOUS VOICE AND DATA SYSTEM**

[75] Inventors: **Gordon Bremer; Kenneth D. Ko**, both of Clearwater; **Luke J. Smithwick**, New Port Richey, all of Fla.

[73] Assignee: **AT&T Corp.**, Murray, N.J.

[21] Appl. No.: **76,530**

[22] Filed: **Jun. 14, 1993**

[51] Int. Cl.^{6} **H04L 5/12; H04L 27/04**
[52] U.S. Cl. **375/261;** 375/259
[58] Field of Search 375/37, 38, 39, 375/59

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Primary Examiner—Sandra L. O'Shea
Assistant Examiner—John Ning
Attorney, Agent, or Firm—Joseph J. Opalach

[57] **ABSTRACT**

In a simultaneous voice and data communications system, a voice signal is mapped into an N-dimensional signal space constellation as a vector, and added to a reference signal point, i.e., a data symbol, where the data symbol is selected from a constellation of data symbols every signaling interval as a function of a data signal. The constellation signal space is further divided into a number of shaped regions, where each region is associated with a different one of the data symbols. The shape of each region is selected as a function of the characteristics of the voice, or audio, signal. In particular, each region illustratively has a rectangular shape.

14 Claims, 5 Drawing Sheets
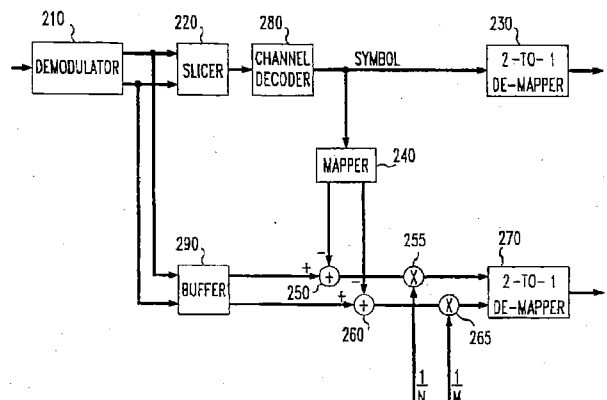
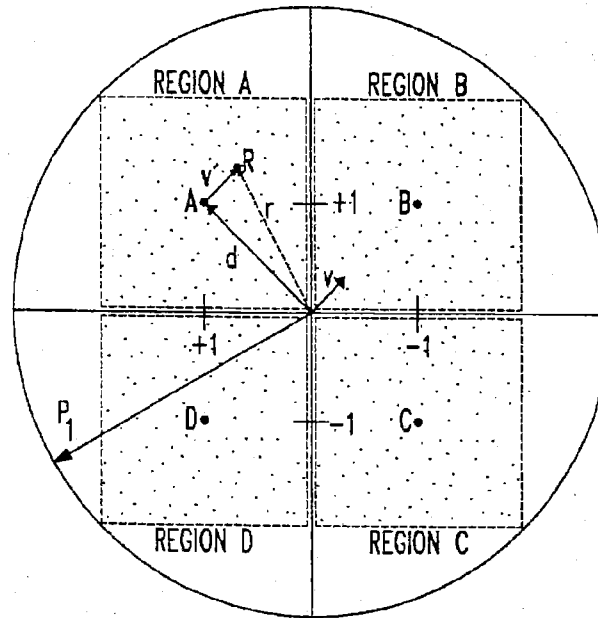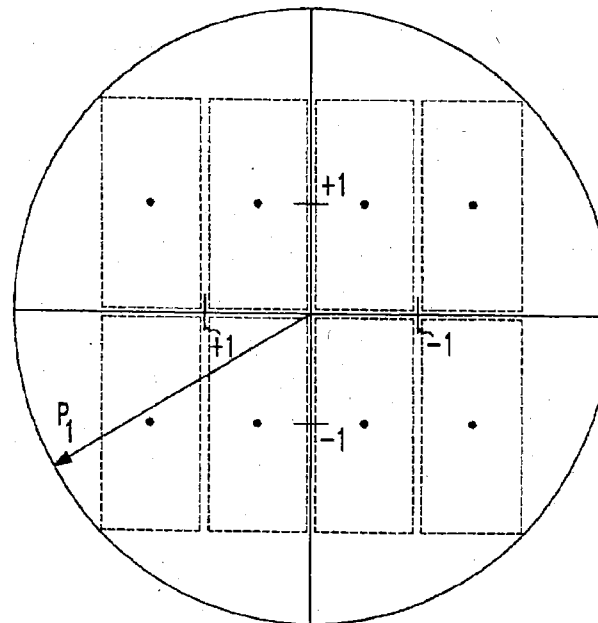**U.S. Patent**   Dec. 12, 1995   Sheet 1 of 5   **5,475,713**

FIG. 1

FIG. 4

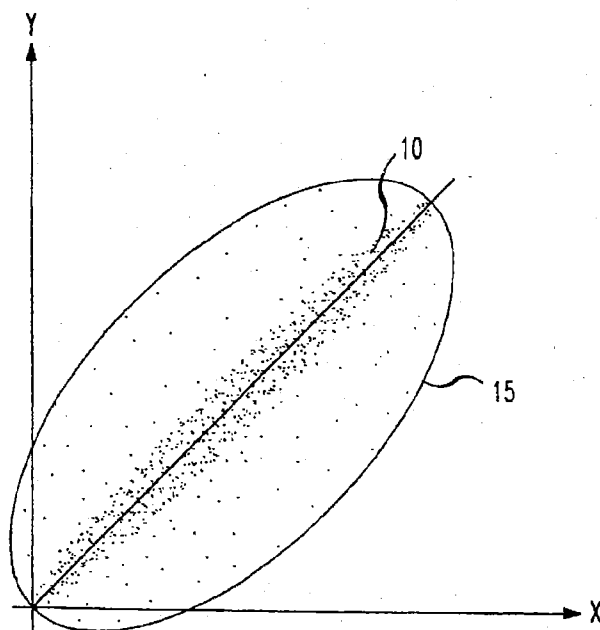**U.S. Patent**   Dec. 12, 1995   Sheet 2 of 5   **5,475,713**

FIG. 2

FIG. 3
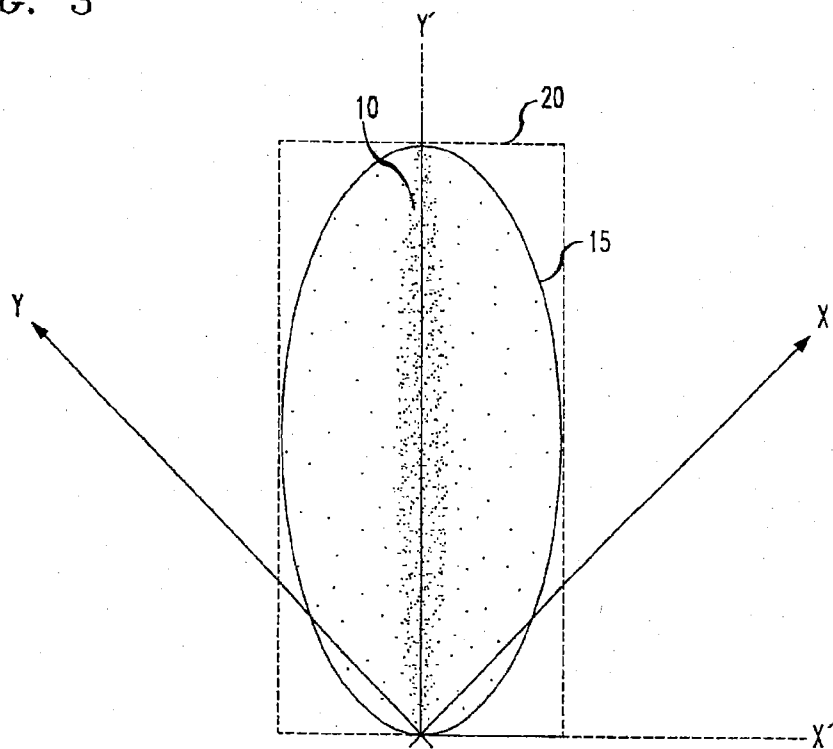
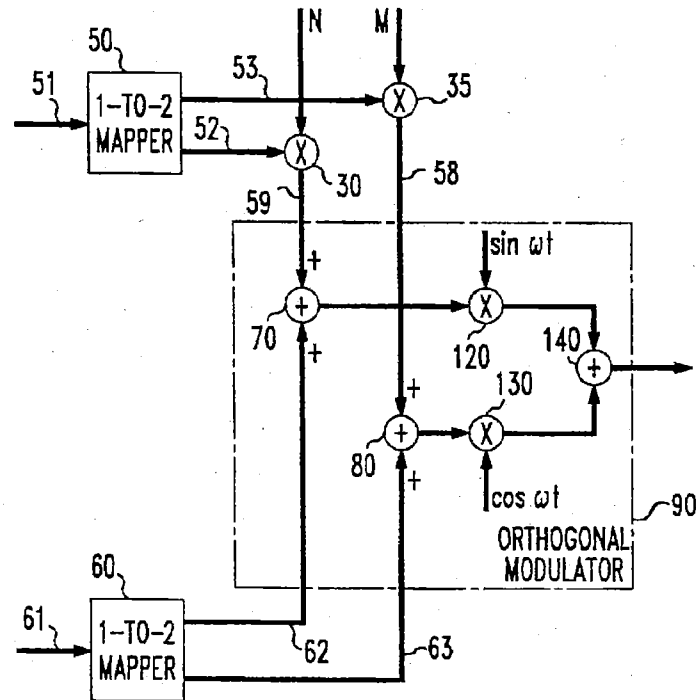**U.S. Patent**   Dec. 12, 1995   Sheet 3 of 5   **5,475,713**

FIG. 5

FIG. 6

FIG. 7

FIG. 8
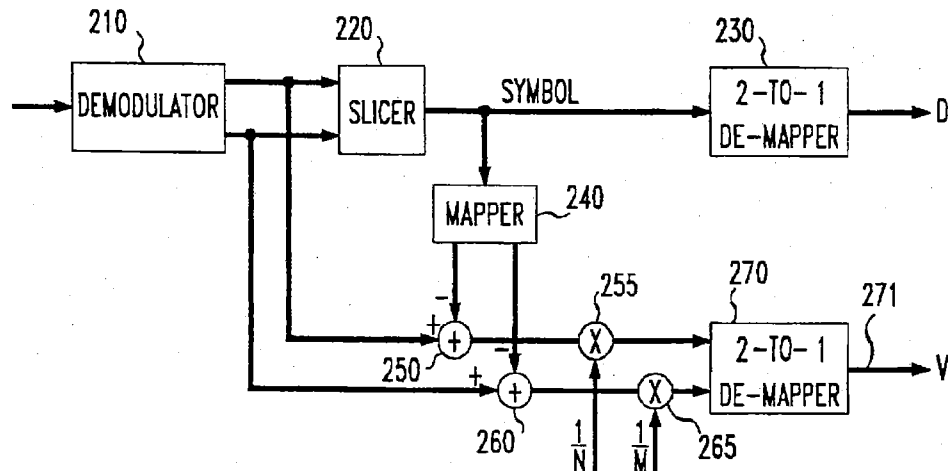
FIG. 9
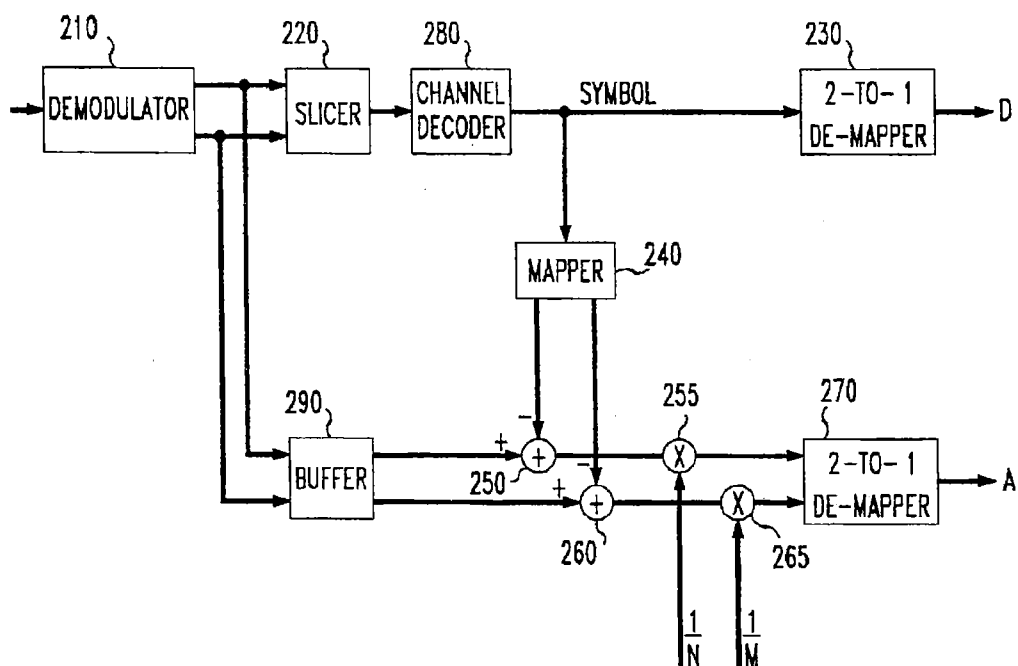
5,475,713

1

SHAPED SIGNAL SPACES IN A SIMULTANEOUS VOICE AND DATA SYSTEM

BACKGROUND OF THE INVENTION

The present invention relates to data communications equipment, e.g., modems. In particular, this invention relates to the transmission of both voice and data signals over the same communications facility at the same time.

The co-pending, commonly assigned, U.S. Patent application of Gordon Bremer and Kenneth D. Ko entitled "Simultaneous Analog and Digital Communication," Ser. No. 08/076,505, filed on Jun. 14, 1993, discloses a simultaneous voice and data communication system in which a voice signal is added to a data signal for transmission over a communications channel to a receiving modem. In particular, the data signal to be transmitted is represented by a sequence of data symbols, where each data symbol is associated with a particular N-dimensional signal point value take from a signal space. Similarly, the voice signal is processed so that it is mapped into the N-dimensional signal space to provide a voice signal point. This voice signal point defines the magnitude and angle of a voice signal vector about the origin of the signal point constellation. The data symbol and the voice signal vector are then added together to select a resultant N-dimensional signal point.

Since this resultant N-dimensional signal point is a function of both the particular data symbol and the magnitude and phase of the voice signal vector, the signal space is essentially divided into a number of square non-overlapping regions, each region comprising a number of signal point values with one signal point value being associated with a data symbol. Consequently, any signal point value within a square region may be validly selected for transmission to a receiving modem. Since each signal point region is non-overlapping, the size of each region is fixed. This results in limiting the magnitude, or dynamic range, of the voice signal vector so that the resultant signal point always lies within a region. The dynamic range of the voice signal vector is directly related to the transmission quality of the voice signal.

Typically the overall size of the signal space is constrained by power limitations. As a result, the modem's data rate and quality of voice transmission are interrelated. For example, for higher voice quality, the signal space is divided into a smaller number of larger size square regions. Each square region is associated with a data symbol and the data rate is directly proportional to the number of data symbols available for transmission. Since there are fewer, but larger, square regions, the dynamic range of the voice signal, and therefore its quality of transmission, increases at the expense of a lower data rate, which is the result of fewer available data symbols. However for lower voice quality, the signal space is divided into a larger number of smaller size square regions. Now since there are more, but smaller, square regions, the dynamic range of the voice signal, and therefore its quality of transmission, decreases with the benefit of a higher data rate. Consequently, any attempt to increase the data rate results in a deterioration in the quality of voice transmission.

SUMMARY OF THE INVENTION

In accordance with the invention, we have realized that the shape, or dimensions, of the signal point regions disclosed in the above-mentioned patent application do not

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have to be square but can be chosen in accordance with the characteristics and processing of the voice signal, or any auxiliary signal, which is simultaneously transmitted with the data signal. As a result, the shape of each region is chosen in such a way as to maximize the number of available data symbols while at the same time maintaining a given level of voice quality. Therefore, the quality of the voice signal is maintained, yet a higher data rate is achieved.

In an embodiment of the invention, a simultaneous voice and data communications system partitions the signal space of a constellation into a number of different regions. The shape of each region is selected as a function of the characteristics of the voice signal. In particular, each region illustratively has a rectangular shape, which provides a better match to the characteristics of the voice signal. By using rectangular shaped regions a higher data rate can be supported for a given level of voice quality since a constellation of signal points can be divided into more regions (and hence support a higher data rate) while still maintaining a given level of quality in the voice transmission. For example, since each region is rectangular, the allowable magnitude of the voice signal is maintained in at least one dimension, while decreasing the magnitude in the other dimension. This allows additional data symbols to be packed into the signal space without changing the dynamic range of the voice signal and concomitantly, maintains the overall quality of the voice transmission at a higher data rate.

BRIEF DESCRIPTION OF THE DRAWING

FIG. 1 shows an illustrative non-overlapping signal point constellation for use in a simultaneous voice and data communications system;

FIG. 2 shows a graph that generally illustrates the characteristics of a two-dimensional voice signal vector over a period of time;

FIG. 3 shows the graph of FIG. 2 rotated by 45 degrees;

FIG. 4 illustrates a signal point constellation used in the transmitter of FIG. 5;

FIG. 5 shows a block diagram of a transmitter embodying the principles of the invention; and

FIG. 6 shows a block diagram of a receiver, which receives the transmitted signal from the transmitter of FIG. 5;

FIG. 7 shows a block diagram of another transmitter embodying the principles of the invention;

FIG. 8 illustrates a signal point constellation used in the transmitter of FIG. 7; and

FIG. 9 shows a block diagram of a receiver, which receives the transmitted signal from the transmitter of FIG. 7.

DETAILED DESCRIPTION

The above-mentioned co-pending patent application of Gordon Bremer and Kenneth D. Ko, which is hereby incorporated by reference, discloses a transmitter in which the transmitted signal represents a sequence of N-dimensional signal points, where each signal point is a function of at least two separate signals. Illustratively, one of these signals is an analog, or voice signal, and the other signal is a data signal. An illustrative 4-QAM symbol constellation for use in the simultaneous voice and data system disclosed in the above-mentioned co-pending patent application is shown in FIG. 1. As described in this co-pending patent application, a data signal is first encoded in each signaling interval, T, to select

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one of four non-overlapping regions, each region represented by one of the reference symbols "A," "B," "C," and "D." For example, the symbol "A" represents region 1, and in particular is associated with the signal point represented by the signal space coordinates (−1, +1), which represents the "in-phase" and "quadrature" components as is known in the art. In a prior art data-only communications system, a transmitter only transmits one of the four symbols selected from this symbol constellation. In other words, there are only four valid signal point coordinates that can be transmitted. However, as disclosed in the co-pending application of Gordon Bremer and Kenneth D. Ko, another signal, e.g., a voice signal, is added to the selected data symbol, with the result that any signal point within a chosen region is selected for transmission. Therefore, each valid region comprises a continuous range of possibly valid signal points and each region is associated with a different data symbol. For example, in every signaling interval, T, the voice signal is processed in such a way to provide two samples of the voice signal. These two samples are used to create a "voice signal vector" in the constellation signal space, e.g., the voice signal vector v shown in FIG. 1. In other words, the two samples of the voice signal, every signaling interval, select a signal point, i.e., a "voice signal point," about the origin of the signal space. The coordinates of this voice signal point define the magnitude and angle of the voice signal vector v from the origin of the signal space. This voice signal vector is than added, i.e., by vector addition, to a data vector, d, which represents the selected data symbol in each signaling interval, T. The resultant vector, r, selects a particular signal point, R, for transmission from the selected region. As can be observed from FIG. 1, the magnitude of the voice signal vector v is constrained so that the resulting signal point, R, lies within a region.

From FIG. 1 it can be seen that each region has a square shape and is centered about the signal point that is associated with each data symbol. Consequently, the magnitude of any voice signal vector is limited in size to one half of the width of each region. Unfortunately, voice signals have a much larger dynamic amplitude range and have an amplitude probability distribution heavily weighted toward zero, i.e., voice is mostly very small in amplitude with only occasional bursts to near full scale. Consequently, any constraint on the magnitude of the voice signal vector directly affects the allowable dynamic range of the voice signal and thereby limits the quality of transmission of the voice signal. For example, it can be observed from FIG. 1 that only four square regions fit within the circle defined by the radius P_1. This circle represents power limitations, which in reality, are imposed on the size of the signal space. Therefore, in order to pack in more data symbols to increase the data rate, the size of each square region must be reduced, which as mentioned above degrades the quality of voice transmission.

As disclosed in the above-mentioned co-pending patent application, the voice vector, v, is formed by sampling a voice signal at twice the symbol rate, 1/T. The voice samples are grouped into pairs which form the coordinates of the voice signal point, which thereby defines the voice signal vector, v. Any voice signal can be viewed as comprising both "voiced" and "unvoiced" components. Generally speaking, the voiced components represent highly correlated speech signals, while the unvoiced components represent uncorrelated speech signals that may include background noise. During sampling of the voiced components of the voice signal, each sample is correlated to the next sample. In other words, consecutive samples will not vary in magnitude to a great degree. However, sampling of the unvoiced compo-

nents will result in more random distribution of voice sample values, therefore consecutive samples will tend to vary in magnitude. This is illustrated in FIG. 2, which generally plots the values of the sample pairs, which form the coordinates of the voice signal point. Since the voice components are correlated, each sample of a sample pair during a voiced interval is generally equal to the other, with the result that the resultant voice signal vector changes in magnitude along line 10 shown in FIG. 2. However, during an unvoiced interval, each sample of a sample pair may be different with the result that there is a deviation from line 10 as shown by area 15 of FIG. 2. Alternatively, this is shown in FIG. 3, where both line 10 and area 15 are rotated by 45 degrees. As can be seen from FIG. 3, the characteristics of a voice signal vector, and the underlying voice signal, are such that the dynamic range requirements in each dimension are different. Indeed, the shape that the voice signal vector, v, traces over a period of time fits within rectangle 20 of FIG. 3 since the more correlated samples are mapped to a longer axis and the less correlated samples are mapped to a shorter axis. Although the above description is in the context of a voice signal, it should be noted that the "sample pairs" of any analog signal, when sampled at a high enough rate, will be highly correlated.

Therefore, and in accordance with the invention, the dimensions of each region of signal space are chosen as a function of the characteristics of the voice signal vector, which in turn is a function of both the voice signal and the method of processing the voice signal. In this example, the resultant shape of each region is a rectangle, which provides a better match to the characteristics of the voice signal samples than the square regions disclosed in the co-pending patent application of Gordon Bremer and Kenneth D. Ko. As a result, a higher data rate can be supported for a given level of voice quality since a constellation of signal points can be divided into more regions (and hence support a higher data rate) while still maintaining a given level of quality in the voice transmission. In other words, by using regions having a rectangular shape, the quality of the voice signal is maintained or increased for a given data rate.

In particular, FIG. 4 shows an illustrative signal point constellation in which each of the regions are rectangular. Compared to FIG. 1, each region of FIG. 4 has one half of the width along the real axis while maintaining the same width along the imaginary axis. As can be seen from FIG. 4, the number of data symbols has been increased from four (shown in FIG. 1) to eight in FIG. 4. In other words, more data symbols are packed into the circle defined by the radius P_1, yet the voice signal's allowable dynamic range, which is mapped into the imaginary direction is the same. As a result, a higher data rate is provided with a higher quality of voice transmission then was heretofore available.

FIG. 5 depicts an arrangement that very basically illustrates the principles of this invention. A voice signal is applied to 1-to-2 mapper 50 via line 51. Mapper 50 samples the voice signal at twice the symbol rate 1/T every signaling interval T, and develops two signals on lines 52 and 53, which are applied to elements 30 and 35 respectively. These two signals represent the coordinates of a voice signal point as shown in FIG. 3. Mapper 50 not only samples the voice signal, but also rotates the samples by 45 degrees, i.e., mapper 50 changes the way the voice signal is mapped into the signal space. In accordance with the invention, elements 30 and 35 scale the output of mapper 50 in such a way that the resultant signal point from the vector addition of the voice vector and the data vector (discussed below) lies within the selected rectangular region. For the purposes of

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this example, M, which represents the real axis, is equal to N/2. In other words, elements 30 and 35 bound the voice vector to the selected region.

In each signalling interval T, encoded digital signals are applied to 1-to-2 dimensional mapper 60, which develops two output pulses on lines 62 and 63, each of which possesses an amplitude that relates to the encoded digital signals arriving on line 61. These two output pulses represent the coordinates in the signal space of one of the data symbols from FIG. 4. Outputs 59 and 62 are combined in adder 70 and outputs 58 and 63 are combined in adder 80. The outputs of adders 70 and 80 form the components of the signals, i.e., the resultant signal point for transmission selected from the signal space of FIG. 4. The outputs of adders 70 and 80 are applied to modulators 120 and 130 and summed in adder 140 to form a modulated signal as is typically known in the art. Although not shown for simplicity, it should be noted that the outputs of adders 70 and 80 can be additionally processed by filters that are bandlimited to no more than ω, in order to avoid aliasing, and to at least half the inverse of the output sample rate of mapper 60.

In FIG. 5 element 60 is depicted as a 1-to-2 mapper. However, it should be understood that this elements can be an M-to-N mapper. That is, element 60 can be responsive to a plurality (M) of encoded digital signals and it can develop a different plurality (N) of output signals. Similarly, element 50 can be an M-to-K encoder that is responsive to a plurality of analog signals. Likewise, the collection of elements that follow elements 50 and 60 (i.e., elements 70, 80, 120, 130 and 140), which form orthogonal modulator 90 can be constructed to be responsive to the plurality of outputs of elements 50 and 60. More specifically, those elements must account for all of the applied input signals, and that means that they must be able to handle K or N signals, whichever is larger. In such a circumstance, however, the user can assume that the larger of the two (K or N) is the dimensionality of the system, and some of the dimensions have either no digital data, or no analog data, whichever applies. Of course, if there are "dimensions" for which there is no digital or analog data, other information can be sent over those dimensions, such as equalization "side" information.

In the context of a signal space, the N pluralities of output signals of elements 50 and 60 (assuming N is larger than K) correspond to the collection of components of vectors in multi-dimensional space; e.g., N-dimensional space. The coordinates of this multi-dimensional space correspond to the orthogonal modulation signals within orthogonal modulator 90. In FIG. 5, the two orthogonal modulation signals are cos ωt and sin ωt, but other modulation signals are also possible. For example, code division multiplexing (CDMA) templates. For purposes of this invention, orthogonal modulation signals are modulation signals that develop a transmitted signal comprising concurrent element signals and yet allow the receiver to separate the received signal into its constituent element signals, those being the signals developed in response to each of the modulation signals. It may also be observed that, relative to FIG. 1 as discussed above, orthogonal modulator 90 performs vector summation of the symbol vector represented by the components developed by element 60 with the analog information vector represented by the components developed by element 50.

FIG. 6 presents a basic block diagram of a receiver in conformance with the principles of this invention. The modulated input signal received from the channel is applied to demodulator 210 which develops the in-phase and quadrature components. Those are applied to slicer 220 which identifies the symbols, i.e., maps the received signal

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point to the closest data symbol from the constellation of FIG. 4. Slicer 220 provides the identified symbols to de-mapper 230, which performs the inverse function to mapper 60 discussed above. In addition, FIG. 6 includes 1-to-2 mapper 240 that is responsive to the symbols developed by channel slicer 220. The output of mapper 240 is the set of in-phase and quadrature components (that are applied in the FIG. 5 arrangement to elements 70 and 80). The outputs of mapper 240 are subtracted from the outputs of demodulator 210 in subtracters 250 and 260. The outputs of subtracters 250 and 260 are applied to elements 255 and 265, respectively. The latter performs the inverse scaling function of elements 30 and 35 discussed above. The output's of elements 255 and 265 are applied to 2-to-1 de-mapper 270 which performs the inverse function of mapper 50 and recombines the voice samples to form the voice signal on line 271.

Another transmitter embodying the principles of the invention is shown in FIG. 7 and illustratively uses the signal point constellation shown in FIG. 8. The transmitter of FIG. 7 is similar to the transmitter of FIG. 5 except for the addition of channel encoder 40, which is responsive to digital signals applied on line 41. Channel encoder 40 operates in accordance with known encoding techniques and is illustratively a trellis encoder like that specified in the CCITT V.32 standard, or shown in other prior art, e.g., U.S. Pat. No. 4,941,154 issued Jul. 10, 1990 to L.-F. Wei. A trellis encoder is a form of convolutional code. In each signalling interval T, channel encoder 40 provides a trellis encoded digital signal to 1-to-2 dimensional mapper 60, which maps the trellis encoded digital signal into the signal space constellation shown in FIG. 8. The remainder of the transmitter functions as described above for the transmitter of FIG. 3.

As disclosed in the co-pending U.S. Patent application of William L. Betts, Gordon Bremer and Edward S. Zuranski, Ser. No. 08/076,507, filed on Jun. 14, 1993 channel encoding the data signal, in a simultaneous voice and data communication system, allows the regions of the signal space to overlap notwithstanding the fact that occasionally "hard errors" are introduced at the receiver. Consequently, for a given region shape, even more dense packing of the data symbols can be achieved. This is illustrated by the signal space shown in FIG. 8. This signal space is similar to the signal space of FIG. 4 except that each region now overlaps with each adjacent region. As can be seen from FIG. 8, the number of data symbols packed into the signal space defined by the circle of radius P_1 is more than the number of data symbols packed in the signal space shown in FIG. 4, yet the quality of the voice signal is the same even though a higher data rate is achieved.

FIG. 9 presents a basic block diagram of a receiver in conformance with the principles of this invention. This receiver is similar to the receiver of FIG. 6 except for the addition of buffer 290, and channel decoder 280. The modulated input signal received from the channel is applied to demodulator 210 which develops the in-phase and quadrature components. Those are applied to slicer 220 which calculates an "error metric" or "transition metric" for each of the data symbols from the constellation of FIG. 8. This error metric is simply the distance the received signal point is from each data symbol. Slicer 220 provides the error metric for each of the data symbols to channel decoder 280, which is illustratively a Viterbi decoder as is known in the art. Generally speaking, channel decoder 280 adds each transition metric to a stored "path metric" for each state of the trellis code and then selects a new path metric for each trellis code state. Each new path metric has the smallest sum

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of error metrics. In other words, channel decoder **280** estimates a received data symbol sequence before deciding on a particular data symbol in order to minimize the overall errors in the entire received symbol sequence. For example, as shown in FIG. **8**, upon receiving signal point R, a hard decision would result in selecting data symbol B. However, this is a hard error since for that one received data symbol, the corresponding error metric is the smallest when compared to the error metrics for the remaining data symbols, and by itself leads to selecting the wrong data symbol. In other words, the voice signal vector v was added to symbol A in such a way that the receiver made a hard error in determining the received symbol. However, channel decoder **280** allows the receiver to recover from, or avoid, this hard error because of the underlying trellis code that was used to select each one of the data symbols for transmission. Consequently, channel decoder **280** provides the corrected data symbol, i.e., symbol A to de-mapper **230** a number of symbol intervals latter, where the number of symbol intervals is typically equal to the decoding depth.

In order to recover the voice signal, FIG. **9** includes buffer **290**, which receives the outputs from modulation **210** and provides a time delay that allows for the trellis decoding to be completed first in channel decoder **280**. FIG. **9** includes a 1-to-2 mapper **240** that is responsive to the symbols developed by channel decoder **280**. The output of mapper **240** is the set of in-phase and quadrature components (that are applied in the FIG. **7** arrangement to elements **70** and **80**). The outputs of mapper **240** are subtracted from the outputs of buffer **290** in subtracters **250** and **260**. The outputs of subtracters **250** and **260** are applied, via inverse scaling elements **255** and **265**, to 2-to-1 de-mapper **270** which recombines the voice samples to form the original voice signal. De-mapper **270** performs the inverse function of mapper **50**.

The foregoing merely illustrates the principles of the invention and it will thus be appreciated that those skilled in the art will be able to devise numerous alternative arrangements which, although not explicitly described herein, embody the principles of the invention and are within its spirit and scope.

For example, although the invention is illustrated herein as being implemented with discrete functional building blocks, e.g., element **30**, etc., the functions of any one or more of those building blocks can be carried out using one or more appropriate programmed processors.

In addition, other shapes can be used, e.g., elliptical shapes, which radiate from the center of the signal space. Also, other processing can be performed on the analog signal to further alter the resultant shape of each region, e.g., differential encoding of the analog signal.

We claim:

1. A method for transmitting comprising the steps of:

a) selecting a data symbol as a function of a data signal, where each data symbol represents a different one of a number of regions of an N-dimensional signal space, and each data symbol is associated with a different N-dimensional signal point in the N-dimensional signal space;

b) selecting an N-dimensional signal point as a function of an applied signal, where the signal point is selected from the N-dimensional signal space;

c) adding the signal space coordinates of the selected data symbol and the selected N-dimensional signal point to provide a resultant signal point;

where each resultant signal point is taken from the region

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associated with the data symbol and where each one of the number of regions has a non-square shape.

2. The method of claim 1 wherein the shape of each one of the number of regions is a function of the characteristics of the applied signal.

3. The method of claim 2 wherein the applied signal is a voice signal and each coordinate of the signal point selected in step b) is a sample of a voice signal.

4. The method of claim 1 wherein the selecting step b) includes the step of:

d) sampling the applied signal to provide a number of samples at most equal to the number of dimensions of the N-dimensional signal space;

e) scaling the number of samples to provide a number of scaled samples, where the scaling step is performed in such a way that at least one of the number of samples is scaled differently from the remaining ones of the number of samples; and

f) selecting the N-dimensional signal point as a function of the number of scaled samples.

5. A method for use in a communications system for transmitting a data signal and an applied signal, comprising the steps of:

defining a number of N-dimensional signal spaces, where each N-dimensional signal space includes a number of regions and where at least two of the N-dimensional signal spaces have a different number of regions and a size of at least one dimension of each region is identical;

selecting one of the number of N-dimensional signal spaces; and

transmitting an N-dimensional signal point from the selected N-dimensional signal space, where the N-dimensional signal point is taken from one of the number of regions of the selected N-dimensional signal space as a function of the data signal and the applied first signal,

wherein the applied signal is a voice signal and the shape of each region is proportional to the dynamic range of the voice signal.

6. The method of claim 5 wherein the number of regions in each one of the number of N-dimensional signal spaces is proportional to a data rate of the communications system.

7. A method for transmitting comprising the steps of:

processing a first signal to provide a sequence of N-dimensional first signal points, each first signal point taken from an N-dimensional signal space, where each first signal point of the sequence defines a respective first signal vector from the origin of the signal space to the first signal point;

processing a data signal to provide a sequence of data symbols, each data symbol associated with a different N-dimensional signal point from the signal space;

adding each first signal vector to a respective data symbol from the sequence of data symbols to prove a sequence of resultant signal points; and

transmitting a signal that represents the sequence of resultant signal points;

where the magnitudes of at least one of the first signal vectors is limited in one dimension more than in another dimension.

8. The method of claim 7 wherein the first signal is a voice signal.

9. Apparatus for transmitting comprising:

first selecting means for selecting an N-dimensional sig-

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nal point as a function of a first signal where the signal point is selected from an N-dimensional signal space;

second selecting means for selecting a data symbol as a function of a data signal, where each data symbol represents one of a number of regions of the signal space, and each data symbol is associated with a different signal point in the N-dimensional signal space; and

means for adding in every signaling interval, T, the signal space coordinates of the two selected signal points to provide a resultant signal point;

where each resultant signal point is taken from the region associated with the data symbol occurring in a respective signaling interval, and where each one of the number of regions has a non-square shape.

10. The apparatus of claim 9 wherein the shape of each one of the number of regions is a function of the characteristics of the first signal.

11. The apparatus of claim 10 wherein the first signal is a voice signal and each coordinate of the signal point selected by the first selecting means is a sample of the voice signal.

12. The apparatus of claim 9 wherein the first selecting means includes:

means for sampling the first signal to provide a number of samples at most equal to the number of dimensions of the N-dimensional signal space;

means for scaling the number of samples to provide a number of scaled samples, where the scaling is performed in such a way that at least one of the number of samples is scaled differently from the remaining ones of the number of samples; and

third selecting means for selecting the N-dimensional signal point as a function of the number of scaled samples.

13. Apparatus for transmitting comprising:

means for processing a first signal to provide sequence of N-dimensional first signal points, each first signal point taken from an N-dimensional signal space, where each first signal point of the sequence defines a respective first signal vector from the origin of the signal space to the first signal point;

means for processing a data signal to provide a sequence of data symbols, each data symbol associated with a different N-dimensional signal point from the signal space;

means for adding each first signal vector to a respective data symbol from the sequence of data symbols to prove a sequence of resultant signal points; and

means for transmitting a signal that represents the sequence of resultant signal points;

where the magnitudes of at least one of the first signal vectors is limited in one dimension more than in another dimension.

14. The apparatus of claim 13 wherein the first signal is a voice signal.

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