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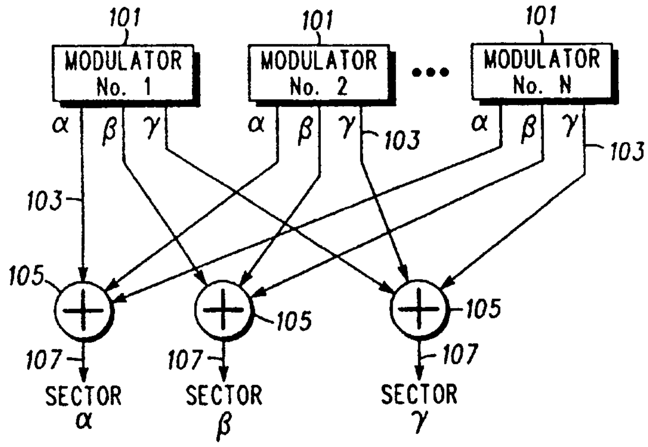

(57) Abstract

A method and apparatus is provided for creating a composite waveform. This composite waveform is created by coding a plurality of input digital information signals (201). Subsequently, the plurality of coded input digital information signals (231, 235, 237) are communicated over a communication medium (103) to a digital combiner (105). The digital combiner (105) combines the plurality of coded input digital information signals. Finally, the digitally combined information signal is spectrally shaped to form a composite waveform (107). These composite waveform creation principals may be applied to digitally encoded voice subbands in a subband coding system as well as channel information in a direct sequence code division multiple access (DS-CDMA) communication system transmitter.

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METHOD AND APPARATUS FOR CREATING A COMPOSITE WAVEFORM

Field of the Invention

The present invention relates to communication systems and, more particularly, to a method and apparatus for creating of a composite waveform.

Background of the Invention

Communication systems take many forms. In general, the purpose of a communication system is to transmit information-bearing signals from a source, located at one point, to a user destination, located at another point some distance away. A communication system generally consists of three basic components: transmitter, channel, and receiver. The transmitter has the function of processing the information signal into a form suitable for transmission over the channel. This processing of the information signal is referred to as modulation. The function of the channel is to provide a physical connection between the transmitter output and the receiver input. The function of the receiver is to process the received signal so as to produce an estimate of the original information signal. This processing of the received signal is referred to as demodulation.

Two types of two-way communication channels exist, namely, point-to-point channels and point-to-multipoint channels. Examples of point-to-point channels include wirelines (e.g., local telephone transmission), microwave links, and optical fibers. In contrast, point-to-multipoint channels provide a capability where many receiving stations may be reached simultaneously from a single transmitter (e.g., cellular radio telephone communication systems). These point-to-multipoint systems are also termed Multiple Access Systems (MAS).

Analog and digital transmission methods are used to transmit an information signal over a communication channel. The use of digital methods offers several operational advantages over analog methods, including but not limited to: increased immunity to channel noise and interference, flexible operation of the system, common format for the transmission of different kinds of information signals, improved security of communication through the use of encryption, and increased capacity.

These advantages are attained at the cost of increased system complexity. However, through the use of very large-scale integration (VLSI) technology, a cost-effective way of building the hardware has been developed.

To transmit an information signal (either analog or digital) over a bandpass communication channel, the information signal must be manipulated into a form suitable for efficient transmission over the channel. Modification of the information signal is achieved by means of a process termed modulation. This process involves varying some parameter of a carrier wave in accordance with the information signal in such a way that the spectrum of the modulated wave matches the assigned channel bandwidth. Correspondingly, the receiver is required to re-create the original information signal from a degraded version of the transmitted signal after propagation through the channel. The re-creation is accomplished by using a process known as demodulation, which is the inverse of the modulation process used in the transmitter.

In addition to providing efficient transmission, there are other reasons for performing modulation. In particular, the use of modulation permits multiplexing, that is, the simultaneous transmission of signals from several information sources over a common channel. Also, modulation may be used to convert the information signal into a form less susceptible to noise and interference.

For multiplexed communication systems, the system typically consists of many remote units (i.e., subscriber units). Each subscriber unit requires a communication channel for short or discrete intervals of time rather than continuous service on a communication channel at all times. Therefore, communication systems have been designed to incorporate the characteristic of communicating with many remote units for brief intervals of time on the same communication channel. These systems are termed time division multiple access communication systems.

A Direct Sequence (DS) Code Division Multiple Access (CDMA) system is a MAS where all subscriber units transmit information signals on the same frequency band simultaneously. Similarly, base stations transmit information signals intended for a particular subscriber unit by transmitting the information signals on the same frequency band as base station originated transmissions to other subscriber units. By necessity, the transmitted bandwidth from a base station is much larger than the information rate of the information signal, i.e. carrier bandwidth is large compared to the message bandwidth.

Most MAS use a digitized representation of voice as the information signal which must be transmitted. This type of information signal is typically produced by a speech coder. It will be appreciated by those skilled in the art that the information signal may also be derived from a data signal such as from a computer modem or data network. A base station typically accepts many information signals as input. In addition, each input signal originates from either a land based telephone or a mobile subscriber unit. Each information signal is multiplexed such that a single modified information signal is transmitted to a base station antenna at a comparatively high data rate. Various stages between the input of the information signals to antenna transmission are susceptible to noise on account of the high data rates involved and the close proximity of signal carrying conductors within the base station. The resulting effect is that capacitive or inductive crosstalk often occurs. As a result, it is desirable to achieve the highest possible utilization of the available carrier bandwidth while minimizing any errors or noise from occurring in the information signal.

Some CDMA system base stations transmit a pilot channel which aids a mobile subscriber in acquiring and tracking a voice channel. A conventional CDMA system provides the pilot channel generating function by a printed circuit board identical to the printed circuit boards used to provide voice and data channels. The output signals of each board is then converted from digital to analog signals. The signals are then combined by a functionally distinct combining circuit.

Conventional CDMA systems route the pilot channel and the voice channels to separate digital to analog converters. This introduces time skew between the pilot channel and the voice channel. If the time skew is large enough, it will create phase inaccuracies in the coherent demodulator of the mobile subscriber unit, thereby decreasing the voice channel performance.

However, even in view of the above-described conventional CDMA communication system, a need still exists for an improved digital information signal combining technique which can be applied to CDMA as well as other types of communication systems.

Summary of the Invention

A method and apparatus is provided for creating a composite waveform. This composite waveform is created by coding a plurality of input digital information signals. Subsequently, the plurality of coded input digital information signals are communicated over a communication medium to a digital combiner. The digital combiner combines the plurality of coded input digital information signals. Finally, the digitally combined information signal is spectrally shaped to form a composite waveform. These composite waveform creation principals may be applied to digitally encoded voice subbands in a subband coding system as well as channel information in a direct sequence code division multiple access (DS-CDMA) communication system transmitter.

Brief Description of the Drawings

FIG. 1 is a block diagram showing a preferred embodiment connection of a Modem Channel Card (MCC) which supplies information to a Baseband Distribution and Combiner (BDC) which then supplies modified information signals to an antenna in accordance with the present invention.

FIG. **2** is a detailed block diagram showing a portion of the preferred embodiment MCC shown in FIG. **1** in accordance with the present invention.

FIG. **3** is a detailed block diagram of another portion of the preferred embodiment MCC shown in FIG. **1** in accordance with the present invention.

FIG. **4** is a detailed block diagram showing the preferred embodiment BDC shown in FIG. **1** in accordance with the present invention.

FIG. **5** is a block diagram showing an alternative preferred embodiment application of digital combining principals to a speech coder in accordance with the present invention.

Detailed Description

The need for an improved digital information signal combining technique is met, as described below, by a method and apparatus which modifies a voice information signal in two stages. The information signal is comprised of data bits which are input to an encoder, which encodes the bits to a higher data rate. Subsequently, forward error correction is introduced to the resultant signal as is data governing the power control of a radio communication subscriber unit transmitter. In addition, the signal is scrambled. Further, an orthogonal Walsh code covers the signal. The first stage culminates in combining the information signal with a number of other information signals from other subscriber units. The resultant signal is then communicated to second stage.

The second stage combines a large number of signals that have originated from an equal number of first stages. The number of signals input to the second stage is large as compared to the number of signals combined in the first stage. Finally, the generated signal is digitally spread and filtered by a band limiting filter. The signals output from the second stage are converted to analog signals. The analog signals are band pass filtered, modulated to the RF carrier, amplified and transmitted by three 120° sector cell antennae for either sector α, β or γ respectively.

By centralizing the band limiting filter as opposed to filtering each channel separately, the total amount of circuitry is reduced and quantization noise is reduced. In addition, since all channels are combined prior to digital to analog conversion, fewer digital to analog converters are necessary. As a final measure, routing a combined voice channel and pilot channel to a single digital to analog converter reduces any time skew between the pilot channel and voice channels inherent in prior art systems that route voice channels through separate digital to analog converters than the pilot channel.

Referring now to FIG. **1**, the system architecture of a three-sector base station may be seen. Three Modem Channel Cards (MCC) **101** are shown. However, it will be appreciated by those skilled in the art that more MCCs (e.g. twenty MCCs) are typically used in a base station. Each MCC **101** takes four digitally encoded voice signals as input. Each voice signal arrives at a rate of 9.6 kilobits per second (Kbps). The MCC **101** performs various operations on the signals and communicates the multiplexed signals **103** to a Baseband Distribution and Combiner (BDC) cards at a rate of 1.288 megasymbols per second (Msym/s). Each BDC **105** accepts several other signals (e.g. 17 signals in addition to the 3 signals shown in FIG. **1**) at a rate matching the rate of the first MCC multiplexed signal.

The BDC **105** performs pseudo-noise (PN) spreading and upsample filters the signal, thereby producing a quadraphase pair of channels **107** suitable for quadrature phase shift keying (QPSK) modulating an analog carrier signal. Further operations occur subsequent to BDC signal processing, such as amplifying the signal. The subsequent signal is Low Pass Filtered and amplified by means well known in the art. The amplified signal is radiated by the one antenna (e.g. sector α) that is operatively coupled to the BDC.

Referring now to FIG. **2** which shows the digital modulator portion **220** of the MCC **101**, digitally encoded voice signals **201** are input to an encoder at a rate of 9.6 Kbps. The encoder increases the data rate by a predetermined factor. In the preferred embodiment, the predetermined factor is two, thereby making the output rate 19.2 Ksym/s. The output from the encoder is in the form of data symbols. The data symbols **205** are then input into an interleaver **207**. Interleaver **207** block interleaves the input data symbols **205**. In the interleaver **207**, the data symbols are input column by column into a matrix and output from the matrix row by row. The interleaved data symbols **213** are output by the interleaver **207** at the same data symbol rate that they were input (e.g., 19.2 ksym/s).

The interleaved data symbols **213** are output to a puncturing algorithm **211**. The puncturing algorithm receives a periodic power adjustment command once every 1.25 milliseconds. The power adjustment command is one or more bits of information. The puncturing algorithm overwrites one or more interleaved data symbols with each power adjustment command. The overwrite by the power adjustment command is treated as an erasure and is corrected by the error correction as decoded in the mobile unit receiver. The resultant interleaved data symbols **213** are output to a scrambler **215**.

The scrambler enhances the security of the communication in the voice channel. The scrambler enhances security by use of a long PN code, exclusive-ORed with the interleaved data symbols. The scrambler outputs the result as scrambled data symbols.

Walsh code spreader **219** provides a unique user code that further spreads the scrambled data symbols. Walsh codes are generated by techniques that are well known in the art. The preferred embodiment uses Walsh codes generated by a Walsh function of order 64. The effect of the Walsh code spreading is to increase the symbol rate by a factor of 64. The Walsh code spreader **219** outputs symbols **221** at a rate of 1.228 Msym/s.

Referring now to FIG. **3** which shows a detailed view of the other portions of the MCC **101** besides the digital modulator portion **220** which was shown in FIG. **2**, gain adjustment commands **223** are inserted into the Walsh code spread signal **221** which provide a subscriber unit with appropriate instructions for subscriber to base site (i.e., uplink) power control. The Walsh code spread signal **221** is multiplied by the appropriate gain for the channel by multiplier **225**. The gain is based on the forward power control level for the channel and the current transmit bit rate. The resultant signal is a multiplied signal **227** consisting of a nine bit signed two's complement integer.

The nine bit signed two's complement integer is input to a parallel to serial converter **229** which produces a single channel serial bit stream **231**. The single channel serial bit stream **231** of the particular voice channel is then input to a serial adder **233**, along with several other serial bit streams **235**, **237** of other voice channels. The preferred embodiment MCC **101** adds four single channel serial bit streams in this manner. Output **239** from the serial adder **233** is split by splitter **241** into antenna sector components α, β, and γ which are communicated across a communication medium (i.e., a base station serial wireline backplane) to the second stage of signal modification as a multiplexed signal. Each multiplexed signal is distributed to a primary BDC **105** which completes the voice channel combining and modulation of the signals in preparation for radio frequency transmission to each of three cell site sectors α, β, and γ. It will be appreciated by those skilled in the art that fewer or more antenna sectors may be implemented without departing from the scope or spirit of the present invention.

Referring now to FIG. **4**, there is shown an α antenna sector BDC **105** which is one of three BDC's used for each base station cell site. Each of the three BDC circuits preferably are implemented on a single printed circuit board to reduce component complexity in the base site. Each BDC circuit corresponds to either sector α, β or γ respectively. A serial modulo-2 adder **301** accepts several multiplexed signals **239** (e.g., 20 α antenna sector signals) as input. In addition, a pilot channel is added to the combined signal **313**. The pilot channel contains no data modulation and is characterized as an unmodulated spread spectrum signal that all the users of a particular cell-site or sector use for acquisition or tracking purposes. In generating the pilot signal, the Walsh "zero" (W_0) sequence which consists of all zeros is used so as not to modulate the pilot signal. Pilot gain register **317** generates a Walsh "zero" (W_0) sequence. The output of the pilot gain register **317** is input to the serial modulo-2 adder **301**. The serial modulo-2 adder **301** outputs a pilot augmented signal **321**. The pilot augmented signal **321** is converted from serial to parallel signals by serial to parallel converter **315**.

Digital Quadrature Phase Shift Key (QPSK) spreading is the next process to be performed. It will be appreciated by one skilled in the art that more or less than four phase shifted signals may be used to modulate the pilot augmented signal **321** (e.g. biphase shift keying (BPSK) could be used). More precisely, any number of phase shifted signals may be utilized in an M-ary phase shift keyed modulating scheme, wherein 'M' denotes the number of phases utilized by the modulating scheme.

Complex QPSK PN spreader **323** provides an In-Phase (I) and Quadrature (Q) channels which are multiplied by binary multiplier **325** with the pilot augmented signal **321**. As a result, two outputs are produced, i.e. the I-channel and Q-channel. Each of the I-channel **327** and Q-channel **329** are filtered by identical bandwidth limiting Finite Impulse Response (FIR) filters **331**. The FIR filters **331** four times upsample the input signal and low pass filter the result to produce two signals: a filtered I-channel **333** and a filtered Q-channel **335**.

Further processing of the signals to produce an amplified RF signal is well known in the art. The filtered I-channel and filtered Q-channel are converted to analog signals. In addition, the filtered I-channel and filtered Q-channel are spectrally shaped by a band pass filter. Subsequently the signals are modulated to the RF carrier, amplified and transmitted by the sector antennae for either sector α, β or γ, respectively.

It will be appreciated by those skilled in the art that the teachings of the present invention may also be applied to provide computationally efficient combining of coded speech information received from different sources. A need for such combining in telephony or land mobile console applications where it is desired to combine separately received user's speech into a single composite that allows all received parties to be heard simultaneously. In telephony, conferencing allows more than two users to simultaneously converse and hear each of the other parties as they speak. In land mobile, a console operator may simultaneously monitor a plurality of radio frequencies using voice and it is necessary to hear all received waveforms superimposed. This capability is also known as N-way conferencing, where N is the number of independent channels combined.

The present invention is particularly applicable to the combining of speech information that is received in coded form. One common speech coder, for example, is the subband coder. One embodiment of such a coder is described, for example, in U. S. Patent 4,979,188 by Kotzin, et. al., which is incorporated herein by reference. It is well known in the art that a subband speech coder conveys information between a transmitter and receiver by coding spectral bands of information separately. In a subband coder, speech samples corresponding to different spectral subbands are separated from the original input and coded for transmission. Such coders employ a plurality of additional techniques, such as noise masking, to improve the quality of the reconstructed speech.

One Figure from the Kotzin et al. '188 patent, included herein as FIG. **5**, shows the preferred embodiment of one such decoder that might be utilized to create one such reconstructed speech output from a single coded received input. Information **502** from a transmission channel is demultiplexed in **504**, splitting the information into various baseband samples associated with separate spectral subbands, and various side information. The side information, among other things, allow the baseband samples to be properly allocated to the appropriate reconstruction filtering to is used to enable the aforementioned improvement techniques. Proper scaling is performed on the samples, and noise samples might be provided to unused subbands. This is accomplished in allocation decoder **538**. The subband samples are directed to the appropriate spectral shaping filters **556-571** via an interpolation stage **540-555**. The spectral shaping may be performed by polyphase anti-aliasing FIR filter banks. The independent subbands outputs are combined to create the reconstructed speech output (**572**).

A straightforward approach to providing N-way conferencing would be to replicate the decoder shown N times, once for each independent input. The outputs could then be simply combined to create a composite waveform.

This alternative preferred embodiment application greatly reduces the computational burden of providing N-way conferencing by this straightforward approach. The basic concept is to combine the multiple speech inputs before the bulk of the computations are performed, namely, before the interpolation and decoder filter bank stages. Each independent channel has its own demultiplexer **504**, which separates the subband samples from the side information. An allocation decoder **538** , in conjunction with received side information, scales and directs the samples to the appropriate spectral subband channel. Additional noise fill samples for other unused subbands might also be provided. The demultiplexer **504** and allocation decoder **538** is shown for only a single voice in FIG. **5**. The side information, among other things, allows proper allocation and amplitude scaling of the filter samples so that the filter samples for this speech input can be properly directed to the interpolation **540-555** and spectral shapers **556-571** to allow speech reconstruction.

However, as shown in FIG. **5**, before the interpolation **540-555** and spectral shaping **556-571**, properly scaled samples from the (N-1) other speech inputs are combined **539** together. This is done separately for each of the possible subbands. Actual speech samples and noise fill may be treated similarly.

Modifications are possible at the intermediate point to optimize the combining for a N-way conferencing scenario. For example, it is possible to scale each of the subband inputs by the energy in its respective signal. This acts to reduce excess residual noise from accumulating but still allows all speech input to be simultaneously combined. The savings using the aforementioned technique are substantial. At most, only one set of decoding filters needs to be provided verses N times that number using traditional techniques.

The principles described herein can be summarized as follows. Referring now to FIG. **1**, a method of creating a composite waveform is shown which includes coding **101** a plurality of input digital information signals. Each input digital information signal may be either digitally encoded voice, data packets, or a combination thereof. Alternatively, each input digital information signal may be derived from a particular subband of a subband coder. Further, each input digital information signal may consist of digital samples representing spectrally partitioned portions of independent information sources (i.e., different voices in a conference circuit). Subsequently, this plurality coded input digital information signals are communicated over a communication medium **103** to a digital combiner **105**. The plurality of coded input digital information signals are digitally combined by the digital combiner **105** and this digitally combined information signal is spectrally shaped to form a composite waveform **107**.

More precisely, a direct sequence code division multiple access (DS-CDMA) transmitter which digitally combines channel information is shown in FIGs. **1-4**. The transmitter includes a first and a second channel encoder **101** for channel coding a first and a second input digital information signal, respectively. Each input digital information signal may include digitally encoded voice, data packets, or any combination thereof. Each channel encoder **101** comprises a splitter **241** for splitting the channel coded digital information signal into sector components α, β, and γ for each sector of a cell. Also, each channel encoder **101** may include a gain controller **225** for adjusting a power gain of the channel coded digital information signal. Further, each channel encoder **101** includes a converter **229** for converting the sector components of the channel coded digital information signal from a parallel to a serial signal. Furthermore, each channel encoder **101** comprises apparati for convolutional encoding **203** , block interleaving **207**, long pseudo-noise code scrambling **215**, and Walsh code spreading **219** each input digital information signal **201**. Finally, each channel encoder **101** may include a mechanism **211** for implementing a puncture algorithm to insert uplink power control information into the digital information signal.

In addition, a combiner **105** is operatively coupled to the first and the second channel encoder **101** for separately serially adding **301** together each sector component of the first and the second channel coded input digital information signals. The combiner **105** includes a mechanism **318** for adding a pilot control channel **317** to the first and the second channel coded input digital information signal. In addition, the combiner **105** comprises a converter **315** for converting each sector component of the digitally combined channel coded digital information signal from a serial to a parallel signal. A modulator **325** is operatively coupled to the converter **315** for separately M-ary phase shift keying each sector component of the digitally combined input digital information signal. In addition, a filter **331** is operatively coupled to the modulator **325** for separately baseband spectrally shaping each sector component of the M-ary phase shift keyed digital information signal to limit the bandwidth of a downlink transmission signal.

The DS-CDMA transmitter preferably further includes an analog transmission portion operatively coupled to the filter **331** which upconverts the M-ary phase shift keyed digital information signal to a radio frequency downlink transmission signal, power amplifies the radio frequency downlink transmission signal, and radiates the amplified radio frequency downlink transmission signal from an antenna.

Although the invention has been described and illustrated with a certain degree of particularity, it is understood that the present disclosure of embodiments has been made by way of example only and that numerous changes in the arrangement and combination of parts as well as steps may be resorted to by those skilled in the art without departing from the spirit and scope of the invention as claimed. For example, the modulator, antennas and demodulator portions of the preferred embodiment communication system as described were directed to CDMA spread-spectrum signals transmitted over a radio communication channel. However, as will be understood by those skilled in the art, the encoding and decoding techniques described and claimed herein can also be adapted for use in other types of transmission systems like those based on time division multiple access (TDMA) and frequency division multiple access (FDMA). In addition, the communication medium (i.e., the radio channel) could alternatively be an electronic data bus, wireline, optical fiber link, satellite link, or any other type of communication channel.

Claims

What is claimed is:

1. A method of creating a composite waveform, characterized by:
   (a) coding a plurality of input digital information signals;
   (b) communicating the plurality coded input digital information signals over a communication medium to a digital combiner;
   (c) digitally combining the plurality of coded input digital information signals with the digital combiner; and
   (d) spectrally shaping the digitally combined information signal to form a composite waveform.

2. A method of digitally combining channel information in a direct sequence code division multiple access (DS-CDMA) communication system, characterized by:
   (a) channel coding a plurality of input digital information signals;
   (b) communicating the plurality channel coded input digital information signals to a digital combiner;
   (c) digitally combining the plurality of channel coded input digital information signals with the digital combiner;
   (d) M-ary phase shift keying the digitally combined input digital information signal; and
   (e) baseband spectrally shaping the M-ary phase shift keyed digital information signal to limit the bandwidth of a downlink transmission signal.

3. The method of claim 2 wherein each input digital information signal is characterized by information selected from the group consisting of digitally encoded voice and data packets.

4. The method of claim 2 wherein the step of channel coding is characterized by at least one of (a) convolutional encoding, block interleaving, long pseudo-noise (PN) code scrambling, and Walsh code spreading each input digital information signal, and

(b) implementing a puncture algorithm to insert uplink power control information into each digital information signal.

5. The method of claim 2 further characterized by at least one of the following steps:
   (a) further comprising the step of splitting each of the plurality of channel coded digital information signals into sector components for each sector of a cell prior to the step of communicating together with the steps of digitally combining, M-ary phase shift keying, and baseband spectrally shaping being separately performed on each sector component;
   (b) the step of adjusting a power gain of the channel coded digital information signal prior to the step of communicating;
   (c) wherein the step of digitally combining being characterized by adding a pilot control channel to the plurality of channel coded input digital information signals; and
   (d) upconverting the M-ary phase shift keyed digital information signal to a radio frequency downlink transmission signal, power amplifying the radio frequency downlink transmission signal, and radiating the amplified radio frequency downlink transmission signal from an antenna.

6. The method of claim 2:
   (a) further characterized by the step of converting each of the plurality of channel coded digital information signals from a parallel to a serial signal prior to the step of communicating;
   (b) the step of digitally combining comprises serially adding together the plurality of channel coded digital information signals; and
   (c) further characterized by the step of converting the digitally combined channel coded digital information signal from a

serial to a parallel signal prior to the step of M-ary phase shift keying.

7. A direct sequence code division multiple access (DS-CDMA) transmitter which digitally combines channel information, characterized by:
   (a) a first and a second channel means for channel coding a first and a second input digital information signal, respectively;
   (b) combiner means, operatively coupled to the first and the second channel means, for digitally combining the first and the second channel coded input digital information signals;
   (c) modulation means, operatively coupled the combiner means, for M-ary phase shift keying the digitally combined input digital information signal; and
   (d) filter means, operatively coupled to the modulation means, for baseband spectrally shaping the M-ary phase shift keyed digital information signal to limit the bandwidth of a downlink transmission signal.

8. The direct sequence code division multiple access (DS-CDMA) transmitter of claim 7 wherein each input digital information signal is characterized by information selected from the group consisting of digitally encoded voice and data packets.

9. The direct sequence code division multiple access (DS-CDMA) transmitter of claim 7 wherein each channel means is characterized by at least one of the following: (a) means for convolutional encoding, block interleaving, long pseudo-noise (PN) code scrambling, and Walsh code spreading the input digital information signal; and (b) means for implementing a puncture algorithm to insert uplink power control information into the digital information signal.

10. The direct sequence code division multiple access (DS-CDMA) transmitter of claim 7 further characterized by at least one of the following:

(a) wherein:

(i) each channel means comprises means for splitting each channel coded digital information signal into sector components for each sector of a cell;

(ii) the combiner means comprises means for separately digitally combining each sector component;

(iii) the modulation means comprises means for separately M-ary phase shift keying each sector component; and

(iv) the filter means comprises means for separately baseband spectrally shaping each sector component;

(b) wherein each channel means comprises means for adjusting a power gain of the channel coded digital information signal;

(c) wherein:

(i) each channel means comprises means for converting the channel coded digital information signal from a parallel to a serial signal; and

(ii) the combiner means comprises means for serially adding together the first and the second channel coded digital information signal and means for converting the digitally combined channel coded digital information signal from a serial to a parallel signal;

(d) wherein the combiner means comprises means for adding a pilot control channel to the first and the second channel coded input digital information signal; and

(e) an analog transmission portion which upconverts the M-ary phase shift keyed digital information signal to a radio frequency downlink transmission signal, power amplifies the radio frequency downlink transmission signal, and radiates the amplified radio frequency downlink transmission signal from an antenna.

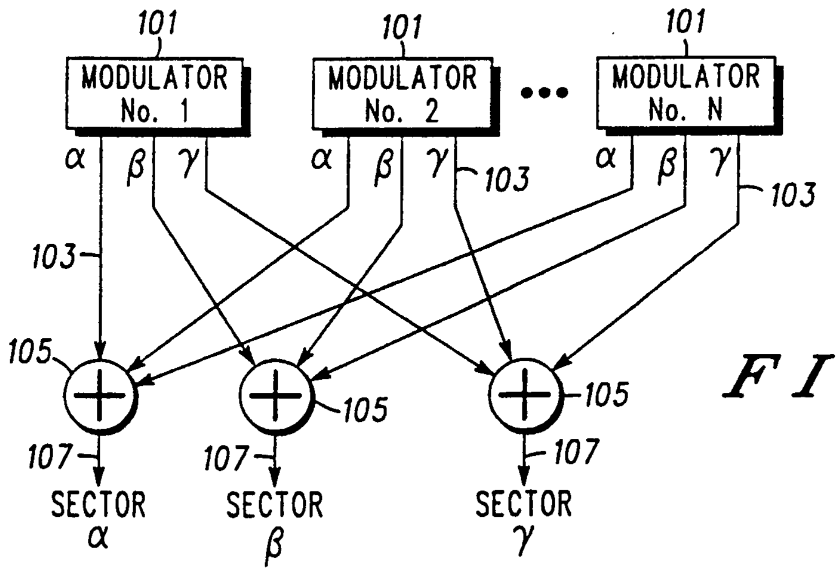


FIG.1

FIG.2

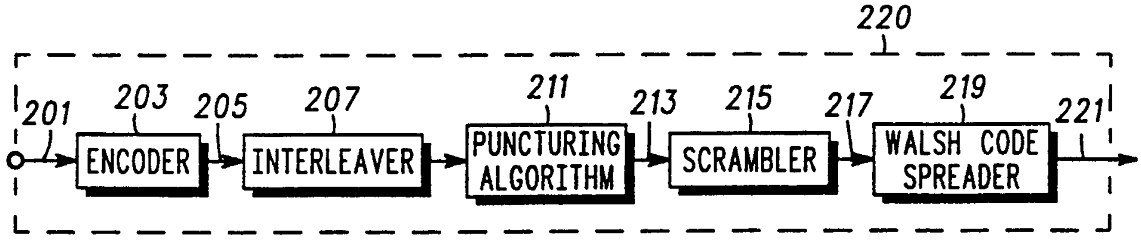
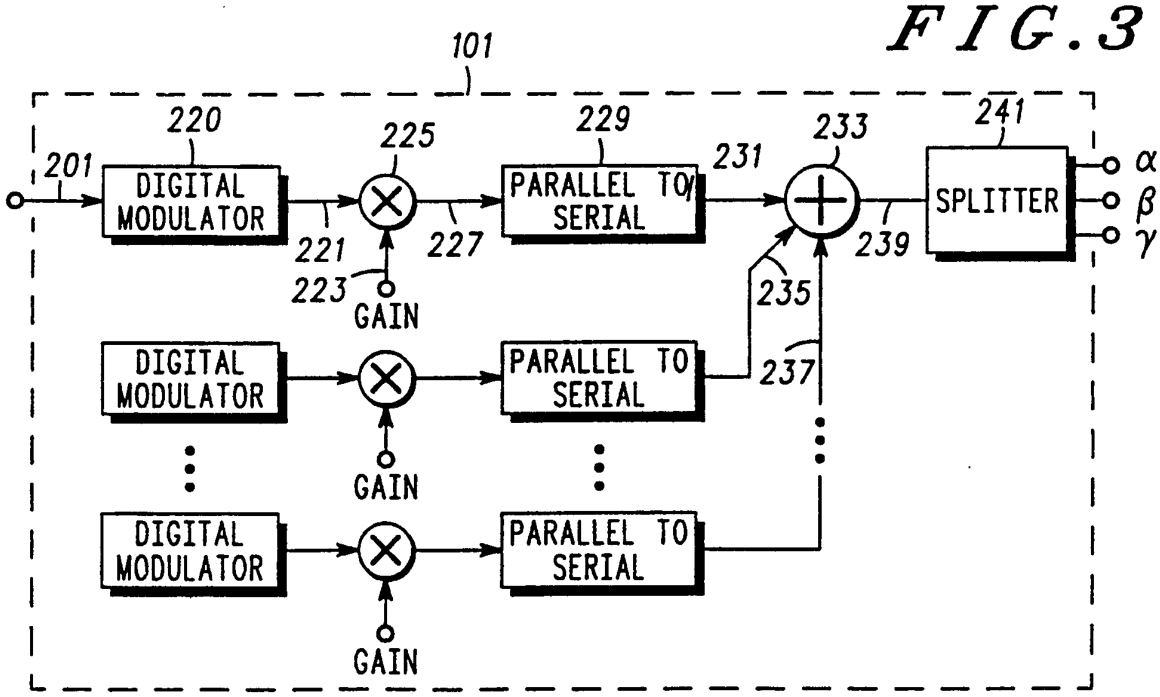


FIG.3



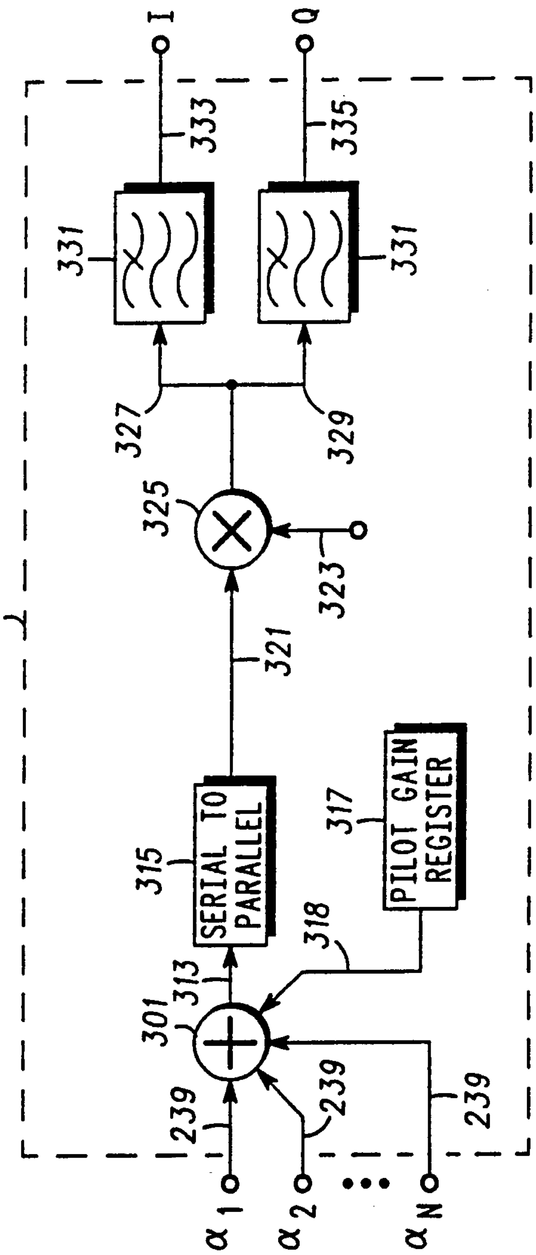

105
SERIAL TO PARALLEL
315
PILOT GAIN REGISTER
317
301
313
318
321
323
325
327
329
331
331
333
335
I
Q
239
239
239
α 1
α 2
α N

FIG. 4

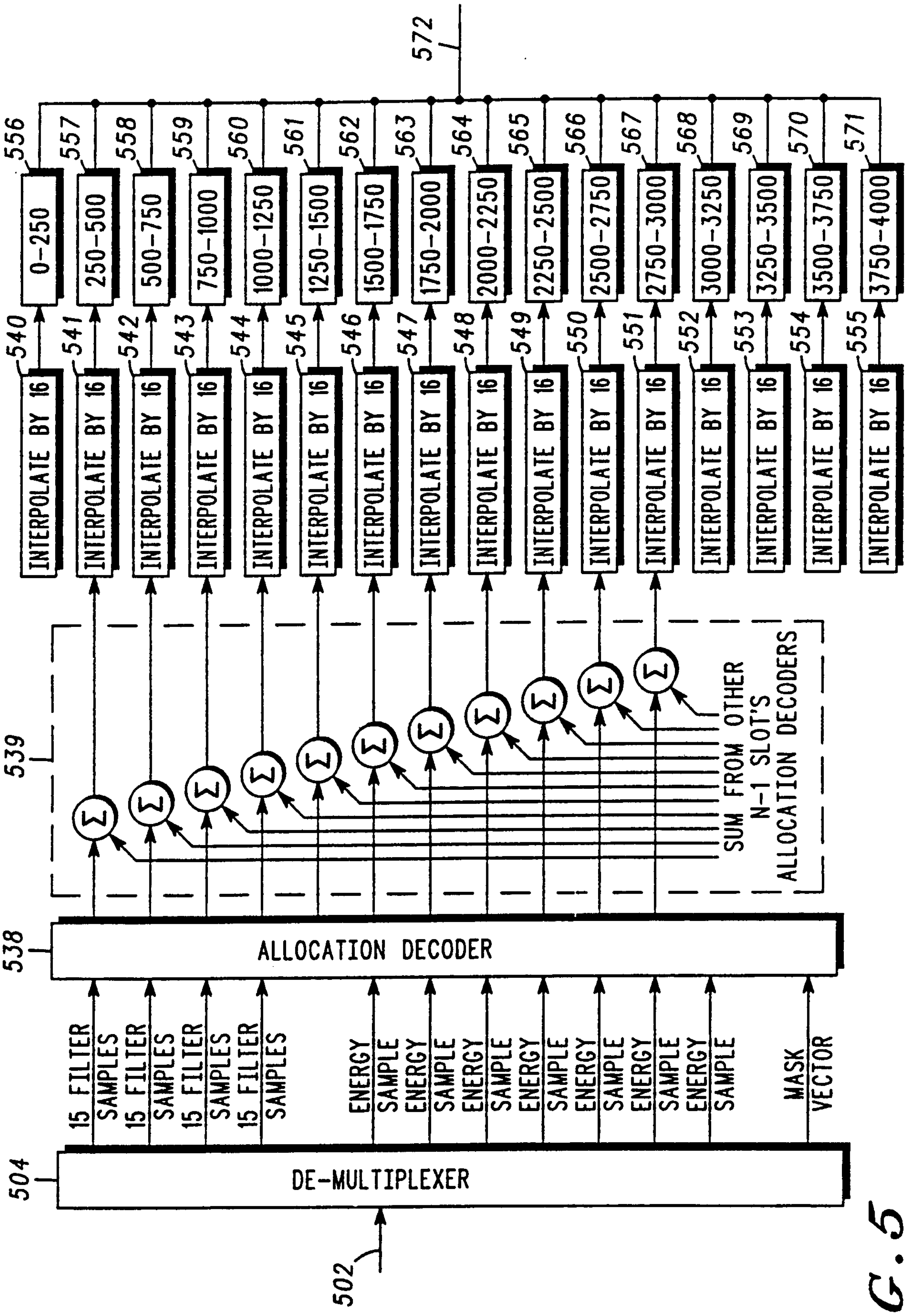

572
556 557 558 559 560 561 562 563 564 565 566 567 568 569 570 571
0-250
250-500
500-750
750-1000
1000-1250
1250-1500
1500-1750
1750-2000
2000-2250
2250-2500
2500-2750
2750-3000
3000-3250
3250-3500
3500-3750
3750-4000
540 541 542 543 544 545 546 547 548 549 550 551 552 553 554 555
INTERPOLATE BY 16
539
SUM FROM OTHER N-1 SLOT'S ALLOCATION DECODERS
538
ALLOCATION DECODER
15 FILTER SAMPLES
ENERGY SAMPLE
MASK VECTOR
504
DE-MULTIPLEXER
502

FIG. 5

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US94/12454

A. CLASSIFICATION OF SUBJECT MATTER
IPC(6) :HO4L 27/30
US CL :375/1
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
U.S. : 375/1
375/27, 40; 370/112

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

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

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US, A, 3,993,867 (BLOOD, JR) 23 NOVEMBER 1976, SEE COLS. 7-10	1
Y	US, A, 4,768,191 (POLCER) 30 AUGUST 1988, SEE FIGS. 3-5.	1
Y	US, A, 5,099,493 (ZEGER ET AL) 24 MARCH 1992, SEE FIGS. 1-3.	1
Y	US, A, 5,103,459 (GILHOUSEN ET AL) 07 APRIL 1992, SEE FIGS. 2-13.	1
A	US, A 5,193,102 (MEIDAN ET AL) 09 MARCH 1993, SEE FIGS. 2-3.	2-10
A,P	US, A 5,289,499 (WEERACKODY) 22 FEBRUARY 1994, SEE FIG. 5.	2-10

[X] Further documents are listed in the continuation of Box C. [ ] See patent family annex.

* Special categories of cited documents:
"A" document defining the general state of the art which is not considered to be part of particular relevance
"E" earlier document published on or after the international filing date
"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&" document member of the same patent family

Date of the actual completion of the international search 18 JANUARY 1995	Date of mailing of the international search report 13 FEB 1995
Name and mailing address of the ISA/US Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231 Facsimile No. (703) 305-3230	Authorized officer Diane Goodwin for SALVATORE CANGIALOSI Telephone No. (703) 308-0482

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT		
Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A, P	US, A, 5,329,547 (LING) 12 JULY 1994, SEE FIG. 1.	2-10

INTERNATIONAL SEARCH REPORT

Box I Observations where certain claims were found unsearchable (Continuation of item 1 of first sheet)

This international report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons:

1. ☐ Claims Nos.:
because they relate to subject matter not required to be searched by this Authority, namely:

2. ☐ Claims Nos.:
because they relate to parts of the international application that do not comply with the prescribed requirements to such an extent that no meaningful international search can be carried out, specifically:

3. ☐ Claims Nos.:
because they are dependent claims and are not drafted in accordance with the second and third sentences of Rule 6.4(a).

Box II Observations where unity of invention is lacking (Continuation of item 2 of first sheet)

This International Searching Authority found multiple inventions in this international application, as follows:

Please See Extra Sheet.

1. ☒ As all required additional search fees were timely paid by the applicant, this international search report covers all searchable claims.

2. ☐ As all searchable claims could be searched without effort justifying an additional fee, this Authority did not invite payment of any additional fee.

3. ☐ As only some of the required additional search fees were timely paid by the applicant, this international search report covers only those claims for which fees were paid, specifically claims Nos.:

4. ☐ No required additional search fees were timely paid by the applicant. Consequently, this international search report is restricted to the invention first mentioned in the claims; it is covered by claims Nos.:

Remark on Protest

☐ The additional search fees were accompanied by the applicant's protest.

☒ No protest accompanied the payment of additional search fees.

BOX II. OBSERVATIONS WHERE UNITY OF INVENTION WAS LACKING
This ISA found multiple inventions as follows:

Group I, claim 1, drawn to a method of creating a digital waveform classified in class 375, subcass 40.

Group II, claims 2-10 drawn to a method and apparatus of Direct Sequence Code Division Multiple access spread spectrum communication classified in Class 375 Subclass 1.
The claims are drawn to two distinct and unrelated inventions. The claims of Group I do not require the particulars of Group II, specifically Direct sequence spread spectrum and phase shift keying and the claims of Group 1 could be used inside a digital computer for communication between the microprocessor and the internal hard drive. Currently, there is no special technical feature; that is, those features which would make the claims avoid the prior art, between Group I and Group II, and Unity is therefore lacking.