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(54) Title: METHODS AND APPARATUS OF PROVIDING TRANSPARENCY OF DTMF SIGNALING TONES ACROSS A DIGITAL CELLULAR VOICE CHANNEL

(57) Abstract

A process is provided for communicating dual tone multiple frequency (DTMF) tone information over wireless connections between encoders and decoders in a digital cellular telephone system comprising the steps of: receiving in an encoder a DTMF tone; identifying in the encoder the received DTMF tone; designating in the encoder a DTMF–encoding frame that digitally encodes the identified DTMF tone; and transmitting the designated DTMF–encoding frame from the encoder to a decoder by radio over a traffic channel of the digital cellular telephone system.

METHOD AND APPARATUS OF PROVIDING TRANSPARENCY OF DTMF SIGNALING TONES ACROSS A DIGITAL CELLULAR VOICE CHANNEL

BACKGROUND OF THE INVENTION

1. Field of the Invention

The invention relates generally to digital cellular telephone communications, and more particularly to the transmission of DTMF tones in digital wireless cellular telephone systems using low-bit-rate speech coders.

2. Description of the Related Art

Dual-tone multiple frequency (DTMF) is a telephone signaling scheme which simultaneously uses one of a group of lower frequency signals and one of a group of higher frequency signals to represent a digit or character. A user of a typical fixed telephone initiates a telephone call over a public switched telephone network (PSTN) by pressing the appropriate dialing key pads to dial the party he or she wishes to call. A conventional fixed telephone incorporates a DTMF tone generator which, under control of the dialing key pad, produces dual-tone signals used to route the call to the called party. The DTMF signals are typically sent over wire connections of a local loop to a local exchange switch of the PSTN which routes the call to the dialed party's telephone.

Wireless cellular telephone systems also employ dialing key pads to input telephone numbers and dialing instructions. However, dialing signals are transmitted in digital messages rather than in DTMF tones, and such digital dialing messages typically are transmitted by actuating a send button on a cellular telephone keyboard. In a wireless cellular telephone system, a radiotelephone, referred to as a subscriber unit, ordinarily communicates with a base station which may be connected to the PSTN. Forward and reverse path radio channels ordinarily are used during radio communications between a subscriber unit and a base station. The forward channel refers to the base-to-subscriber unit path, and the reverse channel refers to the subscriber unit-to-base path. In order to actually deploy a cellular telephone system, a portion of the available radio frequency spectrum must be allocated for use by the system. Unfortunately, there is a scarcity of available radio frequency spectrum, and this scarcity can impose significant limitations on the design of wireless cellular systems, and in particular, on the use of DTMF tones.

More generally, initial deployments of cellular telephone systems used FM analog modulation for sending speech signals. Some advantages of FM modulation have been improved signal quality when compared AM modulation and reduced complexity when compared with digital modulation. Basically, FM modulation expands the bandwidth of a signal to increase redundancy so that unwanted noise can be more readily removed. The tradeoff for improved voice quality is increased bandwidth per user. However, with a scarcity of available frequency spectrum, increasing bandwidth per user can significantly reduce the volume of calls that an FM system can handle.

In recent years, the advent of lower cost digital signal processors (DSPs) has made digital modulation of voice signals more cost effective and practical. As a result there has been a proliferation of digital cellular telephone systems: GSM (worldwide), IS-54/136 (North America), PDC (Japan), IS-54 (North America) and IS-95, for example. These digital cellular systems allow many users to share bandwidth that previously would have been allocated to a single user under an FM analog system. See, D.M. Balston and R.C.V.

Macario, editors, Cellular Radio Systems, Artech House, Inc. 1993, for a discussion of the origin and design of typical cellular telephone systems.

Low-bit-rate digital speech coders implemented using digital signal processors have made digital cellular telephone systems a commercially viable reality. A typical conventional state of the art speech coder transforms a speech waveform from the analog domain to the digital domain, and then compresses the digital bit stream down to between 4.8 kbps and 13 kbps (depending on the system). Generally, a lower bit rate means reduced signal quality. Often, in order to achieve lower bit rates, and correspondingly higher system call handling capacity, speech compression techniques are employed which decompose a signal into components that can be quantized for transmission with fewer bits. Unfortunately, during the compression process, non-speech signals such as DTMF tones may be distorted to the point that they are not transmitted with acceptable fidelity. Thus, a DTMF signal ordinarily cannot pass "transparently" through a digital cellular telephone system.

There are many possible causes of DTMF tone distortion in the course of transmitting such tones over a low bit rate digital cellular telephone system. For example, digital speech encoder ramp-up time may be too long, and as a result, tones may be missed because the encoder is too slow to converge to an output that has acceptable DTMF characteristics. Also, there may be distortion due to bad frame masking, and the high bit error rate of an RF channel may cause DTMF tones to be improperly reconstructed. Additionally, there may be amplitude distortion because the amplitude of a reconstructed speech waveform may vary with time causing one continuous tone to be reconstructed as a sequence of many of the same digit repeated. Finally, twist distortion may occur. Since DTMF uses two frequencies that must have similar amplitudes, digits may be missed when a speech coder attenuates one tone more than the other.

Despite limitations upon the ability of digital cellular telephone systems to handle DTMF signals, there is an increasing need to digitally transmit DTMF signals to digital cellular telephone traffic channels the channels on which actual voice or data is transmitted. Typical digital cellular telephone systems have a multiplicity of traffic channels. For instance, in addition to dialing a phone number, DTMF tones also may be used for caller id (typically outside the U.S.), entering credit card information, and interacting with voice mail systems. Unfortunately, there have been shortcomings with prior approaches to transmitting DTMF tones through digital cellular telephone systems.

For example, U.S. Patent No. 4,922,517, issued to West, et al., entitled, System for Interfacing a Standard Telephone Set with a Radio Transceiver, discloses a wireless cellular telephone system which converts DTMF tones produced by a conventional telephone into digitized versions of the DTMF tones. However, the disclosed system transmits digitized DTMF during call set up. It does not disclose the transmission of actual DTMF tone information over the traffic channel in the course of an actual call, to access voice mail, for instance.

Moreover, there are digital cellular telephone handsets that do not process actual DTMF tones at all. Rather, a microprocessor (often a DSP) in the handset directly generates digital signals that represent DTMF tones, although no actual DTMF tones are produced. These digital wireless telephone systems thereby avoid the use of actual DTMF tones in the combined handset-subscriber unit. More specifically, for example, a hand held subscriber unit may include both an analog voice transceiver and a digital key pad. The microprocessor (the voice coder) in the subscriber unit encodes voice signals. Input to the key pad, for use in caller id, credit card number input or voice mail interaction, for example, also is encoded by the microprocessor which may construct a FACCH (Fast Associated Control Channel) message that represents the key pad input. The FACCH signal can be used by a base station or a switch controller connected to the PSTN to produce a conventional analog DTMF tone for transmission over conventional analog telephone circuits. In this manner, for example, a digital cellular telephone handset can be used to access a voicemail system which is on the PSTN and which is responsive to DTMF tones; even though no actual analog DTMF tone is received or generated by the subscriber unit handset itself.

The use of a microprocessor, such as a DSP, in the telephone handset, however, may not be commercially practical, for example, in wireless local loops in which it may be desirable to employ a conventional low cost handset to transmit voice, data and DTMF signals to a subscriber unit for digital wireless transmission to a base station. These handsets are far less complex and far less expensive than typical hand held subscriber units employed in digital wireless cellular telephone systems. Moreover, these conventional handsets, which are ubiquitous, do in fact produce DTMF tones. Consequently, the deployment of a commercially practical wireless local loop, for example, is best implemented in a digital wireless scheme which can handle analog DTMF tones used by conventional handsets.

Thus, there has been a need to employ conventional telephone handsets that transmit and receive DTMF tones over the traffic channel after call set up in digital wireless telephone systems. The present invention meets this need.

SUMMARY OF THE INVENTION

In one aspect, the present invention provides a process for communicating dual tone multiple frequency (DTMF) tone information over wireless connections between encoders and decoders in a digital cellular telephone system. A DTMF tone is received in an encoder. The received DTMF tone is identified in the encoder. A DTMF-encoding frame is designated to digitally encodes the identified DTMF. The designated DTMF-encoding frame is transmitted from the encoder to a decoder by radio over a traffic channel of the digital cellular telephone system.

Another aspect of the invention provides a novel encoder for use in a digital cellular telephone system. The encoder includes an interface to an analog telephone and includes a radio tuneable to a traffic channel of the digital cellular telephone system. A speech coder produces digital speech coder signals in response to analog voice signals received on the telephone interface. A DTMF detector detects DTMF tones received on the telephone interface. A DTMF-encoding device produces DTMF-encoding frames in response to DTMF tones received on the telephone interface. A selection device alternatively provides the speech coder signals or the DTMF-encoding frame to the radio tuned to the traffic channel.

Yet another aspect of the invention provides a new decoder which includes an interface to an analog telephone and a radio tuneable to a digital cellular telephone system traffic channel. A device produces analog speech signals in response to speech coder signals received on the radio tuned to the traffic channel. A device also produces a DTMF tone in response to a DTMF-encoding frame received on the radio tuned to the traffic channel. A selection device alternatively provides the analog speech signals or the DTMF tone on the telephone interface.

Still another aspect of the invention provides a novel digital cellular telephone system. An encoder in the system includes an interface to an analog telephone and a radio tuneable to a traffic channel of the digital cellular telephone system. A speech coder in the encoder produces digital speech coder signals in response to analog voice signals received on the telephone interface. A DTMF detector detects DTMF tones received on the telephone interface. A DTMF-encoding device in the encoder produces DTMF-encoding frames in response to DTMF tones received on the telephone interface. A seectin device alternatively provides the speech coder signals or DTMF-encoding frames to the radio tuned to the traffic channel. A decoder in the system includes an interface to an analog telephone and a radio tuneable to the traffic channel. The decoder also includes a speech coder which produces analog speech signals in response to digital speech coder signals received on the radio tuned to the traffic channel. The decoder further includes a DTMF tone generator which produces DTMF tones in response to receipt of one or more DTMF-encoding frames received on the radio tuned to the traffic channel. Moreover, the decoder includes a selection device which alternatively provides the analog speech signals or the DTMF tone on the telephone interface.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 provides an illustrative generalized block diagram of an encoder of a digital cellular telephone system in accordance with a presently preferred embodiment of the invention;

Figure 2 provides an illustrative generalized block diagram of a decoder of a digital cellular telephone system in accordance with a presently preferred embodiment of the invention;

Figures 3A and 3B illustrate representative conventional frame structures of reverse and forward traffic channel slots in accordance with the PDC standard;

Figure 4 illustrates representative conventional frame structure of a traffic channel slot in accordance with the GSM standard;

Figures 5A and 5B illustrate representative conventional frame structures of reverse and forward traffic channel slots in accordance with the North American standard;

Figures 6A - 6E are illustrative timing diagrams in which an exemplary analog input signal is shown in Figure 6A, a corresponding DTMF detector signal internal to the encoder of Figure 1 is shown in Figure 6B, corresponding speech coder output frames internal to the encoder are shown in Figure 6C, corresponding traffic channel frames transmitted by the encoder are shown in Figure 6D, and an exemplary analog output signal produced by the decoder of Figure 2 is shown in Figure 6E;

Figure 7 provides an illustrative block diagram showing actual components employed, in a presently preferred embodiment of the invention, to implement a subscriber unit having the encoder features of Figure 1 and the decoder features of Figure 2;

Figure 8 provides an illustrative block diagram showing actual components employed, in a presently preferred embodiment of the invention, to implement a base station and radio switch controller having the encoder features of Figure 1 and the decoder features of Figure 2; and

Figure 9 provides an illustrative generalized block diagram of an alternative decoder of a digital cellular telephone system in accordance an alternative embodiment of the invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

The present invention comprises a novel method and apparatus for transmitting digital DTMF-encoding signals in a digital wireless cellular telephone system. The following description is presented to enable any person skilled in the art to make and use the invention. Descriptions of specific applications are provided only as examples. Various modifications to the preferred embodiment will be readily apparent to those skilled in the art, and the general principles defined herein may be applied to other embodiments and applications without departing from the spirit and scope of the invention.

Thus, the present invention is not intended to be limited to the embodiment shown, but is to be accorded the widest scope consistent with the principles and features disclosed herein.

The presently preferred embodiment of the invention is implemented in a digital cellular telephone system of the general type disclosed in commonly assigned U.S. Patent Application Serial No. 08/400,169, filed March 7, 1995, entitled, Method and Apparatus to Improve P~~P~~STN. Access to Wireless Subscribers Using a Low Bit Rate System, invented by Haoui et al. which is expressly incorporated herein by this reference. It will be appreciated, however, that the principles of the invention can be practiced in other low bit rate digital cellular telephone systems as well. Thus, for example, although the aforementioned commonly assigned patent application discloses a system that conforms with the PDC digital cellular standard, the invention alternatively can be implemented in digital cellular telephone systems compliant with the GSM, IS-54/136 or IS-95 standards.

Referring to the illustrative drawing Figure 1, there is shown a functional block diagram representation of a digital cellular telephone system encoder 20 in accordance with a presently preferred embodiment of the invention. An encoder of the general type shown in Figure 1 may be employed both in both subscriber units and in base stations or switch controllers of a digital cellular telephone system. The encoder 20 includes an interface 21, a microprocessor-based speech coder 22, an FEC (Forward Error Correction) module 24, a digital radio 26 and an antenna 28. The encoder 20 also includes a DTMF detector 30, a controller 32 and a "DTMF-encoding" FACCH (Fast Associated Control Channel) generator 34. The encoder further includes first and second multiplexer circuits 36 and 38.

Referring to the illustrative drawing of Figure 2, there is shown a functional block diagram representation of a digital cellular telephone system decoder 40 in accordance with a presently preferred embodiment of the invention. A decoder of the general type shown in Figure 2 may be employed in both subscriber units and in base stations or switch controllers of a digital cellular telephone system. The decoder 40 includes an antenna 42, a digital radio 44, an FEC module 46, a microprocessor-based (de)coder 48, a multiplexer 50 and an interface 52. The decoder 40 also includes a FACCH detector module 53, an $LAPD_m$ module 54, controller 55, a FSM (Finite State Machine) 56, a DTMF generator 58 and a "Silence" generator 60.

For an understanding of additional details of the presently preferred embodiment that are not directly related to the present invention see the above-identified commonly assigned patent application. It will be understood that the same physical components may serve both as part of the encoder 20 and as part of the decoder 40.

In operation, the encoder 20 shown in Figure 1 may receive speech signals and DTMF signals via the interface 21. In the present embodiment, the interface 21 is a four wire interface. In the case where the encoder 20 is part of a subscriber unit, speech and DTMF signals may be produced by a conventional telephone handset (not shown) connected directly to the subscriber unit. In the case where the encoder 20 is part of a base station, speech and DTMF signals may be produced by a conventional telephone set connected to a PSTN. (not shown). Input signals received on the interface 21 are provided to both the microprocessor-based speech coder 22 and the DTMF detector 30. The speech coder 22 digitally encodes the received speech signals and provides such digitally encoded signals to the first multiplexer 36. Signals output by the speech coder 22 shall be referred to as "speech coder" signals. During call set-up and call-release and possibly at other times, the controller 32 generates "normal" FACCH signals, as defined in the PDC standards specification, which are provided to the first multiplexer 36. The DTMF detector 30 monitors the interface 21. When no DTMF tone is detected, the second multiplexer 38 passes the encoded speech or normal FACCH messages to the FEC module 24 for forward error correction coding. The ECC protected encoded signals then are transmitted via the digital radio 26 and the antenna 28. When a DTMF tone is detected, the DTMF detector 30 signals the DTMF-encoding FACCH generator 34 which produces a DTMF-encoding FACCH (hereinafter, "FACCH/DTMF") message.

More particularly, during the receipt of a DTMF tone, the controller 32 causes the second multiplexer 38 to select the FACCH/DTMF messages produced by the FACCH/DTMF generator 34. These FACCH/DTMF messages are provided to the radio FEC module 24 for ECC protection. It will be appreciated that the controller 32 can alternatively be implemented using the same microprocess (or DSP) employed to produce the speech coder 22. The ECC protected FACCH/DTMF signals then are transmitted to a decoder (in a base station or subscriber unit) via digital radio 26 and the antenna 28.

In the presently preferred embodiment of the invention, a continuous stream of FACCH/DTMF messages may be produced during the receipt of the DTMF tone by the detector 30. Specifically, upon detecting a DTMF tone, the DTMF detector 30 instructs the FACCH/DTMF generator 34 to begin producing digital FACCH/DTMF messages that represent the detected analog DTMF tone. The generator 34 responds by producing a continuous sequence of FACCH/DTMF frames. These frames include a digital code which indicates which DTMF tone has been detected by the DTMF detector 30. In the presently preferred embodiment, there is a different digital code for each DTMF tone.

When the DTMF detector 30 no longer detects the DTMF tone, it instructs the FACCH/DTMF generator 34 to produce special digital end-of-DTMF tone messages that signal the end of the tone. In the presently preferred embodiment, these special end-of-DTMF tone messages comprise two

FACCH/SILENCE messages which are transmitted to signal the end of the tone. After these two final FACCH/SILENCE messages are sent, the controller 32 instructs the second multiplexer 38 to pass normal speech and normal FACCH messages. Note that these messages are termed FACCH/"SILENCE", because, although they are part of the signal sequence used to transmit digitized DTMF tone information, they do not carry information about audible tones but rather indicate the end of the FACCH/DTMF frame sequence. In a present embodiment the FACCH/SILENCE messages may be coded so as to produce actual silence in a receiving decoder or to produce audible "noise" (static) so that a listener does not mistakenly conclude that the wireless connection has been dropped. In the current implementation, if these end-of-DTMF tone messages are missed, then the FSM will automatically generate 40 msec of silence between the DTMF tone and the subsequent voice signal.

In operation, the decoder 40 shown in Figure 2 receives encoded traffic signals via the antenna 42 and the digital radio 44. The received signals are provided to the FEC 46 module which corrects errors in the received signals. The error corrected signals are provided to the microprocessor-based speech coder 48, the FACCH detector 53 and the FSM 56. The speech coder 48 decodes the received "speech coder" signals so as to produce analog voice signals which are provided to the multiplexer 50. The multiplexer 50, which is controlled by the FSM 60, provides the analog voice signals to the interface 52. In a present embodiment, the interface 52 is a four-wire interface. If the decoder 40 is part of a subscriber unit, then the interface 52 may be connected to a conventional telephone or telephone answering machine, for instance. This connection may be a conventional wire connection or a conventional cordless telephone connection. If on the other hand, the decoder 40 is part of a base station, then the interface 52 may be connected to the PSTN, for example. Normal FACCH messages are detected by the FACCH detector 53 and are processed by the controller 55 which, for example, may control hand-offs, radio-channel measurements, maintenance and supplementary services, all of which will be understood by those skilled in the art. These FACCH messages typically "steal" time slots from the voice or data traffic channels in order to communicate irregular control requirements such as call handover, for example. The $LAPD_m$ module 54 allows for retransmission. Basically, retransmission ensures that DTMF digits are received in the proper sequence and that none are missing. The $LAPD_m$ is a varient of LAPD specified in the ITU, Q.921 standard which will be understood by those skilled in the art. Briefly, the $LAPD_m$ algorithm implemented in a presently preferred embodiment of the invention allows for addressing using a variable length field, and retransmission using a standard sliding window "go back N" (GBN) technique. It will be appreciated that there are many possible varients of the retransmission algorithm. For instance, "Selective Repeat" (SR) could be employed instead of GBN.

An advantage of sending a continuous stream of FACCH/DTMF messages during the provision of the DTMF tone is that speech encoding is supplanted, overwritten or "blanked", and therefore, cannot interfere with transmission of digitized DTMF information. Thus, the encoder 20 can flood the voice (or traffic) channel with FACCH/DTMF signals that essentially blank any speech encoded DTMF signals.

An advantage of sending two, rather than one, FACCH/DTMF message at the end of the tone is redundancy in case one FACCH/DTMF message is missed. In the current embodiment, the first two FACCH/DTMF messages are accorded higher priority in the system than regular traffic, such that the controller 32 is permitted to override the FACCH/DTMF messages with its own normal FACCH messages only after the first two FACCH/DTMF messages have been sent. Similarly, two FACCH/SILENCE messages are transmitted not only for redundancy in indicating the end-of-DTMF tone, but also to meet certain DTMF tone specifications. It will be understood by those skilled in the art that there are DTMF specifications that prescribe a silence interval following a DTMF tone, although it is not uncommon to ignore this silence requirement.

Furthermore, although the present embodiment of the invention is implemented in accordance with the PDC standard, the principles of the invention apply to other digital cellular telephone standards as well. Accordingly, the following paragraphs describe traffic channel frame structure not only according to the PDC standard, but also according to other exemplary digital wireless cellular telephone systems such as, the GSM, IS-54 and IS-95 standards as well.

Referring to the illustrative drawing of Figures 3A and 3B, there is shown the conventional organization of exemplary time slots within reverse and forward traffic channels in accordance with the PDC standard. The guard time is 6 bits (G) and corresponds to the maximum cell radius of 20km. An additional 4 bits (R) are assigned to suppress interference caused by up and down ramp burst amplitudes. The preamble is 2 bits (P). The traffic (TCH), which can be encoded voice, data or FACCH is a total of 224 bits divided into two 112 bit blocks. A 20 bit synchronization word (SW), used as a training word for multipath equalization, is placed in the middle of the slot. An 8 bit "color code" (CC) is used to identify the current base station. A 1 bit steal flag (SF) is used to discriminate TCH from FACCH. It will be appreciated that, in accordance with the PDC standard, TCH information may be interrupted by normal FACCH message which are used to used communicate irregularly occurring control messages. The 15 bit Slow Associated Control Channel (SACCH) carries control information such as receiving level and interference level, which are used for radio link control during the call.

The illustrative drawings of Figure 4 provides an example of conventional traffic channel frame structure of a traffic slot for encoded voice or data in accordance with the GSM standard. Each time slot lasts 0.577 ms and comprises 148 bits with an 8.25 bit guard period between slots. The GSM standard also employs FACCH messages to impart irregular control requirements. The traffic carried by the slot is divided into two separate 57 bit blocks. Each block is assigned data from separate speech coding frames. Eight such blocks are required to convey 20ms of speech data, but each slot carries data from two different speech blocks simultaneously. One control bit associated with each data block is used to indicate whether the block is carrying normal traffic or has been stolen by the FACCH. In the center of the traffic slot is a sequence of 26 bits used by the receiver to set the parameters of equalizer and demodulator in order to overcome multipath problems, for example.

The illustrative drawings of Figures 5A and 5B provide an example of conventional traffic channel frame structure of a traffic slot for encoded voice or data reverse and forward channels in accordance with the IS-54 North American standard. There are three traffic slots per channel, one for each of three subscriber units that can communicate over the channel at a time. An entire channel frame has a length of 20 ms. Hence, the length of each traffic slot within a frame is 20/3 ms. A 6 bit guard time is provided to prevent adjacent time slots from colliding in a base station due, for example, to variations in propagation time to the base station for different subscriber units sharing the channel. A 6 bit ramp time is allocated to permit smooth turn on and turn off of the subscriber unit. A 28 bit synchronization and training (SYNC) word is provided. A 12 bit Slot Associated Control Channel (SACCH) is provided. A 12 bit Coded Digital Verification Color Code (CDVCC) is provided. The DVC is an 8 bit identifier used to separate subscriber units that are on the same physical channel, but controlled by different nearby base stations, so-called co-channels. The DVCC is protected by a shortened Hamming code to produce the 12 bit CVDCC. The reverse channel slot carries two 122 bit DATA blocks that may contain data or

FACCH messages. The forward channel slot carries two 130 bit DATA blocks that may contain data or FACCH messages. There are 12 bits of reserved (RSVD) in the forward channel. The FACCH is used for urgent control signals such as handover orders, and when an FACCH message is to be transmitted, it replaces a speech information in a DATA block.

Under the IS-54 North American standard, each 159 bit speech block is error protected using a process in which different classes of bits in the block are provided with different levels of error protection. In contrast the 49 bits in a typical FACCH message all receive the same level of error protection. For a forward channel slot, a DATA block will comprise 260 bits whether it carries error protected speech or an error protected FACCH message. No flag is provided in the traffic slot to distinguish speech and FACCH content. Rather, the DATA block content is ascertained based upon the difference in error coding employed for speech and FACCH blocks.

For a more complete discussion of the exemplary digital cellular telephone standards discussed above, including further details of the different channel coding schemes under the different standards, refer to, D.M. Balston and R.C.V. Macario, editors, Cellular Radio Systems, chapters 6, 9 and 10, Artech House, Inc. 1993.

Thus, although the current implementation of the invention employs the PDC FACCH frame structure, other FACCH frame structures, such as those used in GSM or IS-54 for instance, could be employed in other implementations of the invention. Moreover, it should be understood that although the presently preferred embodiment employs FACCH frames to represent digital DTMF-encoding signals, the invention is not intended to be limited to the use of FACCH frames. Any digital DTMF-encoding pattern may be employed which is distinguishable from normal speech or normal FACCH messages, for example. Ideally, such alternative DTMF-encoding would provide for some form of error protection.

The provision of DTMF-encoding signals over the traffic channel of a digital cellular telephone system in accordance with the presently preferred embodiment of the invention will now be explained with reference to the illustrative drawings of Figures 6A-6E and the "Decoder Side FSM State Transition Table 'A'". Figures 6A-6E show an exemplary analog input signal in Figure 6A, a corresponding DTMF detector signal internal to the encoder 20 in Figure 6B, corresponding speech coder output frames internal to the encoder 20 in Figure 6C, corresponding traffic channel frames transmitted by the encoder 20 in Figure 6D and an exemplary analog output signal of the decoder 40. The Decoder Side FSM State Transition Table "A" provides a state transition diagram which explains the operation of the finite state machine (FSM) of the decoder 40.

Referring to Figure 6A, there is shown an exemplary analog input signal 70 provided to the interface 21 of the encoder 20. The input signal includes a first (analog) speech signal 72, a DTMF tone signal 74 and a second (analog) speech signal 76. Figure 6B illustrates the response of the DTMF detector 30 to receipt of the DTMF tone 74: a digital representation 81 of the DTMF tone which is delayed by approximately 40 msec because of the time typically required by a detector of the type employed in the present embodiment to detect a valid tone. The digital produced by the DTMF detector 30 is provided to the DTMF-encoding FACCH generator 34.

Meanwhile, as illustrated in Figure 6C, the speech coder 22 produces internal (to the encoder 20) digital speech coder frames in response to the analog input signal 70. First, speech coder (coded speech) frames 82 are produced by the speech coder 22 in response to the first speech signal 72. These first speech coder (coded speech) frames 82 collectively represent a digitized version of the first speech signal 72. The second speech coder (coded silence) frames 83 are produced by the speech coder 22 in response to the silence, between the first speech signal 72 and the DTMF tone signal 74. These second speech coder frames 83 represent a digitized version of that silence. The third speech coder (coded DTMF) frames 84 are produced by the speech coder 22 in response to the DTMF tone signal 74. These third speech coder (coded DTMF) frames 84 represent a digitized version of the DTMF tone 74. The fourth speech coder (coded silence) frames 85 are produced by the speech coder 22 in response to the ending of the DTMF tone signal 74. The ending of DTMF tone 74 is manifested by a silent interval between the tone 74 and the beginning of second speech signal 76. These fourth speech coder (coded silence) frames 85 represent a digitized version of that silence. Finally, fifth speech coder (coded speech) frames 86 (only one frame shown) are produced by the speech coder 22 in response to the second speech signal 76. These fifth speech coder frames 86 collectively represent a digitized version of the second speech signal 76.

In the present embodiment, there is an approximately 48 msec delay between the receipt of any given component of the analog signal illustrated in Figure 6A and the production of a corresponding frame(s), shown in Figure 6C, by the speech coder 22. This represents the time required by the DSP to run the speech coding algorithm using the received analog signal as input data.

Figure 6D illustrates an exemplary sequence of frames that are actually outputted by the encoder 20 in response to the analog signals in Figure 6A. In the presently preferred embodiment, these frames are a combination of speech frames produced by the speech coder 22, normal FACCH frames produced by the controller 32 and special (FACCH/DTMF frames or FACCH/SILENCE frames) produced by the FACCH/DTMF generator 34. In the current embodiment, the FACCH/DTMF generator can produce two types of frames explained above: FACCH/DTMF and FACCH/SILENCE. The speech coder

22 digitally encodes analog signals received on the interface 21. The controller 32 may periodically produce normal FACCH frames which interrupt the flow of speech frames in a conventional manner via the first multiplexer 36. The generator 34 produces special FACCH frames (FACCH/DTMF or FACCH/SILENCE) which are inserted into the traffic flow via the second multiplexer 38 upon the detection of a DTMF tone.

More specifically, the encoder 20 outputs the first digital speech frames 92 produced by the speech coder 22 (subject to ECC protection) that correspond to the first analog speech signal 72. The encoder also outputs the second digital silence frames 93 (subject to ECC protection) that correspond to the silence between the first analog speech signal 72 and the analog DTMF tone 74. Then, immediately after a DTMF tone is detected by the DTMF detector 30, at least two FACCH/DTMF frames 94-1 are produced by the FACCH/DTMF generator 34 and are inserted into the flow of traffic frames via the second multiplexer 38. The two inserted FACCH/DTMF frames 94-1 each contain a digital representation or encoding which identifies which DTMF tone has been detected. When the DTMF detector 30 detects the end of the DTMF tone, the generator 34 produces at least two FACCH/SILENCE frames 94-3 and inserts these into the sequence of traffic frames.

In the presently preferred embodiment of the invention, during the time interval between the provision of the last of the first two FACCH/DTMF frames 94-1 and the provision of the first of the first two FACCH/SILENCE frames 94-3, normal FACCH frames (none shown) may be produced by the controller 32 and may be inserted into the traffic sequence. In the presently preferred embodiment, however, during this interval no other coded frames, (coded speech, coded silence or coded DTMF) are inserted into the traffic sequence because that could cause the state machine to end DTMF operation and return to voice recovery. Thus, although speech is blanked, the controller

32 may cause the multiplexer 36 to insert normal FACCH frames into the sequence of traffic frames during the interval when the DTMF tone is detected.

In the current embodiment, the multiplexers 32 and 36 are implemented in a microprocessor which sets a bit when a first FACCH/DTMF message is encountered and does not allow other normal FACCH messages until the bit is cleared in response to the second FACCH/DTMF message. If the controller 32 does not seek to introduce normal FACCH frames into the traffic flow then the special FACCH generator 34 inserts an uninterrupted sequence of error protected FACCH/DTMF frames 94-2, representing the detected DTMF tone, into the traffic sequence until the DTMF detector 30 detects the end of the tone. An advantage of this repeated insertion of FACCH/DTMF frames is precise control of length of the DTMF pulse, during error-free operation. When the end of the tone is detected, the two FACCH/SILENCE frames 94-3 are inserted. After that, in accordance with the example, speech frames 96 (only one shown) that correspond to the second analog speech signal 76 and the fifth coded (coded speech) frames 86 are output.

Referring to the illustrative drawing of Figure 6E, there is shown an exemplary analog output signal 100 produced on the interface 52 of the decoder 40 in response to the exemplary sequence of digital traffic frames, shown in Figure 6D. A first regenerated analog speech signal 102 corresponds to the first speech signal 72. A regenerated analog DTMF tone signal 104 corresponds to the DTMF tone signal 74. A second regenerated analog speech signal 106 corresponds to the second speech signal 76.

In operation, the decoder 40 receives the exemplary digital traffic frames illustrated in Figure 6D via the antenna 42 and digital radio 44. The received frames are provided to the FEC module 46 for error detection and correction. The output of the FEC module 46 is provided to the FACCH detector 53, FSM 56 and speech coder 48. The controller 55 responds to normal FACCH traffic frames in a conventional manner that is well known to those skilled in the art. The $LAPD_m$ module provides for retransmission as will be understood by those skilled in the art. The speech coder 48 converts error corrected digital traffic frames to analog speech signals which are provided to the multiplexer 50. The FSM 56 controls the operation of the FACCH/DTMF generator 58 and the FACCH/SILENCE generator 60 and the multiplexer 50. More particularly, the FSM 56 evaluates the traffic frames output by the FEC module 46, and determines whether to output on the interface 52 either: analog speech/silence signals decoded by the speech coder 48 or a DTMF tone generated by the DTMF generator 58 or a SILENCE signal generated by the SILENCE generator 60.

Thus, referring to both Figures 1 and 2, when a DTMF tone is transmitted in the midst of ordinary analog speech signals, for example, frames encoded by the speech coder 22 are blanked by the FACCH/DTMF and FACCH/SILENCE frames produced by the generator 34. The FSM 56, in response to this sequence of frames, controls the selection of decoded (analog) speech signals produced by the speech coder 48, "silence" produced by silence generator 60 or an analog DTMF tone produced by the DTMF generator 58.

The operation of the FSM 56 in controlling the regeneration of analog DTMF tones from digitized versions of the DTMF tones transmitted on the traffic channel now will be described with reference to the following exemplary Decoder Side FSM State Transition Table "A".

<u>Decoder Side FSM State Transition Table "A"</u>

State	Input	Next State	Output
Speech	FACCH/DTMF	DTMF	DTMF
Speech	FACCH/SILENCE	SILENCE 1	SILENCE
Speech	Other	Speech	Speech

DTMF	Speech	SILENCE 1	SILENCE
DTMF	FACCH/SILENCE	SILENCE 1	SILENCE
DTMF	Other	DTMF	DTMF
SILENCE 1	all	SILENCE 2	SILENCE
SILENCE 2	Speech	Speech	Speech
SILENCE 2	FACCH/DTMF	DTMF	DTMF
SILENCE 2	Other	SILENCE 2	SILENCE

An important role of the FSM 56 is to handle situations where frames are missed due to errors on the traffic channel or normal FACCH frames intermixed with FACCH/DTMF. The FSM 56 also ensures that the DTMF is off for 40ms between digits as required in typical digital cellular system specifications.

The FSM 56 of the presently preferred embodiment exhibits four states: Speech, DTMF, SILENCE1 and SILENCE2.

When the FSM 56 is in the Speech state, and a digital FACCH/DTMF traffic frame is received, then the FSM 56 transitions to the DTMF state and causes the multiplexer 50 to select the output of the DTMF tone generator 50. The DTMF tone generator 58 produces a DTMF tone corresponding to the DTMF tone digitally encoded by the received FACCH/DTMF frame. Accordingly, the decoder output is the DTMF tone.

When the FSM 56 is in the Speech state, and a digital FACCH/SILENCE frame is received, then the FSM transitions to a SILENCE1 state and causes the multiplexer 50 to select the output of the silence generator 60. Thus, the decoder outputs an analog SILENCE signal. This transition to the SILENCE1 state helps to ensure that there is a 40ms interval between DTMF tones since, as explained below, the next state transition will be to

SILENCE2 state, and each of SILENCE1 and SILENCE2 is 20ms in duration. Specifically, the expected operation of the encoder 20 is to always insert two FACCH/SILENCE frames into the traffic channel upon the termination of a received DTMF tone. Moreover, the expected operation of the encoder 20 is to only insert FACCH/SILENCE frames following FACCH/DTMF frames and never following speech frames. Therefore, the receipt of a FACCH/SILENCE frame while the FSM is in a Speech state indicates that FACCH/DTMF frames have been dropped. Hence, in order to ensure at least 40ms of silence between the missed FACCH/DTMF frame and another (possible) FACCH/DTMF frame, the FSM transitions to SILENCE1.

When the FSM 56 is in the Speech state, and any other type of frame (other than FACCH/DTMF or FACCH/SILENCE) is received then the FSM transitions to the Speech state and causes the multiplexer 50 to select the output of the speech decoder 48. That is, there is no state transition. Another type of frame, other than a speech frame, could for example be a normal FACCH frame. The receipt of another speech traffic frame or a normal FACCH frame when the FSM is in the Speech state is consistent with the expected operation of the encoder 20 and the decoder 40. Hence, no state transition.

When the FSM 56 is in the DTMF state and a Speech frame is received then the FSM transitions to the SILENCE1 state and causes the multiplexer 50 to select the output of the SILENCE generator 60. Thus, the decoder outputs SILENCE. The FSM was in the DTMF state because a FACCH/DTMF frame was previously received. The expected operation of the encoder 20 is that once an FACCH/DTMF frame has been transmitted, a speech frame will never be transmitted before the transmission of two FACCH/SILENCE frames. Two FACCH/SILENCE frames are employed to provide redundancy in case one of them is missed due to excessive bit errors for example. Thus, when the FSM is in a DTMF state, and a speech frame is received, then the FSM assumes that the expected FACCH/SILENCE frame has been missed and transitions to SILENCE. Hence, in order to ensure at least 40ms of silence between the previously received FACCH/DTMF frame and another (possible) FACCH/DTMF frame, the FSM transitions to SILENCE1.

When the FSM 56 is in the DTMF state and a FACCH/SILENCE frame is received, then the FSM transitions to the SILENCE1 state and causes the multiplexer 50 to select the output of the SILENCE generator 60. The receipt of FACCH/SILENCE frame when the FSM is in the DTMF state is consistent with the expected operation of the encoder 20 and the decoder 40.

When the FSM 56 is in the DTMF state and any other type of frame is received (other than speech or FACCH/SILENCE) then the FSM transitions to the DTMF state and causes the multiplexer 50 to select the output of the DTMF generator 58. That is, there is no state transition. Another type of frame, for example, could be a normal FACCH frame or a FACCH/DTMF frame. The receipt of any of these "other" types of traffic frame when the FSM is in the DTMF state is consistent with the expected operation of the encoder 20 and the decoder 40.

When the FSM 56 is in the SILENCE1 state, and any traffic frame is received then the FSM transitions to the SILENCE2 state and causes the multiplexer 50 to select the output of the SILENCE generator 60. The expected next frame is a FACCH/SILENCE frame. However, when the FSM is in the SILENCE1 state, then the next state should be SILENCE2 in order to ensure at least a 40ms interval between DTMF tones. If the next received frame is not FACCH/SILENCE then that frame is assumed by the FSM to have been missed. Hence, regardless of the nature of the next received frame, the FSM transitions to SILENCE2.

When the FSM 56 is in the SILENCE2 state and a speech frame is received, then the FSM transitions to the Speech state and causes the multiplexer 50 to select the output of the speech coder 48. The receipt of a speech frame when the FSM is in the SILENCE2 state is consistent with the expected operation of the encoder 20 and the decoder 40.

When the FSM 56 is in the SILENCE2 state and a FACCH/DTMF frame is received, then the FSM transitions to the DTMF state and causes the multiplexer 50 to select the output of the DTMF generator 58. The receipt of a FACCH/DTMF frame when the FSM is in the SILENCE2 state is consistent with the expected operation of the encoder 20 and the decoder 40.

When the FSM 56 is in the SILENCE2 state and any other type of traffic frame (other than speech or FACCH/DTMF) is received then the FSM transitions to the SILENCE2 state and causes the multiplexer 50 to select the output of the SILENCE generator 60. That is, there is no transition. Another type of frame could, for example, be a normal FACCH frame or a FACCH/SILENCE frame. The only other state from which the FSM transitions into the SILENCE2 state is the SILENCE1 state, and the FSM transitions from SILENCE1 to SILENCE2 regardless of the type of frame received next. However, once in the SILENCE2 state, the next expected frame under normal operation of the encoder 20 is either a speech frame or a normal FACCH frame or a FACCH/DTMF frame.

It will be appreciated that the FSM 56 in the present embodiment is implemented as a "Mealy" machine in which output is a function input plus current state. Alternatively, an FSM consistent with the invention could be implemented as a "Moore" machine in which output is a function of the state only. There also are many alternative techniques for handling dropped frames and for ensuring appropriate time intervals between different types of frames.

Thus, the invention is not intended to be limited to the use of any particular state machines in general or to any particular state machine.

Referring to the illustrative drawings of Figure 7, there is shown a block diagram showing components used to implement a subscriber unit 300 in a presently preferred embodiment of the invention. An encoder 20, as described in Figure 1, implemented in a subscriber unit employs a Toshiba TC35305F 302 as the DTMF detector 30. A Motorola 68LC302 304 serves as the controller 32 and is programmed to generate normal FACCH frames. The 68LC302 also is programmed to serve as the special FACCH generator 34 and to generate the special (FACCH/DTMF and FACCH/SILENCE) FACCH frames. The 68LC302 304 is further programmed to serve as the second multiplexer 38. In operation, the TC35305F 302 interrupts the 68LC302 304 in order to call up the programming required to produce the appropriate special FACCH frames in response to the detection of a DTMF tone. An Asahi Kasei Microelectronics AKM 2370/1/3 chip set 306, 307, 308, respectively, is programmed to serve as the first speech coder 22, multiplexer 36, FEC module 24 and digital radio 22. It will be appreciated, for example, that the DTMF detector 30, the second multiplexer 38 and the special FACCH generator 34 can be equivalently implemented using the same DSP that is used to implement the speech coder 22. An echo canceler 310 is implemented using a Texas Instruments TMS320C17, and codec and filters 312 are implemented using a Oki Semiconductor MSM7543. The operation of these components and the other components illustrated in Figure 7 will be readily understood by persons of ordinary skill in the art and will not be further described herein.

Referring to the illustrative drawings of Figure 8, there is shown a block diagram showing components used to implement a base station 400 and radio switch controller (RSC) 402 in a presently preferred embodiment of the invention. An encoder 20, as described in Figure 1, is implemented in the radio and switch controller (RSC) VOX (voice) processors 114-1. The VOX processors 114-1 of the current embodiment are implemented using general purpose DSPs. Specifically, the entire encoder 20 is implemented in firmware on the same DSP as the speech coder: a Texas Instruments TM320C50 Digital Signal Processor. The operation of these and other components of the base station 400 and RSC 402 is explained in the above-identified commonly assigned patent application.

The special FACCH frames are produced using a Layer 2, SAP =1 message whose contents are one byte containing the ASCII code the DTMF digit or blank (0x20) for silence. Thus, coding for the messages was the following hex bytes: C2 02 23 XX, where XX is one of 30, 31, 32, 33, 34, 35, 36, 37, 38, 39, 41, 42, 43, 44, 2A, 23, 20. These codings apply in the presently preferred embodiment which is implemented in accordance with the PDC standard. Different codings might be employed in an implementation according to the GSM, IS-54 or IS-95 standards, for example.

Referring again to Figure 7, in the presently preferred embodiment of the invention, a decoder 40, in accordance with Figure 2, implemented in a subscriber unit 300 employs a Motorola 68LC302 304 programmed to serve as the controller 54, FSM 56 and to implement the $LAPD_m$. An AKM 2373 308 is programmed to serve as the speech coder 48, FEC module 46 and FACCH detector 53. A Texas Instruments TMS320C17 Digital Signal Processor 310 is programmed to serve as the multiplexer 50, DTMF generator 58 and silence generator 60.

Referring again to Figure 8, in a base station 400 and radio switch controller (RSC) 402 in accordance with the presently preferred embodiment, the decoder 40, in accordance with Figure 2, is implemented in the VOX in the RSC described above. The FSM 56 and the DTMF generator 58 are implemented in computer programs using standard programming techniques. The $LAPD_m$ is implemented by programming a 68LC302 118-2 in the RSC.

The DTMF generators in DSPs use a digital resonator to generate sine waves at the appropriate frequencies at a level of -14dBm according to the ITU specification for DTMF generation.

Referring to the illustrative drawings of Figure 9, there is shown a block diagram of an alternative embodiment of a decoder 200 in accordance with the invention. The decoder 200 of Figure 9 may be employed with the encoder 20 of Figure 1. The decoder 200 includes a digital radio 202, an FEC module 204, a speech decoder 206, a SILENCE generator 208 an FSM 210, a FACCH detector 212, an LAPDm module 214, a controller 216, a DTMF generator 218 and first and second multiplexers 220 and 222. The decoder 200 receives digital radio transmissions via antenna 224 and provides decoded outputs on interface 226. The basic functions and implementations of most of the constituent elements of the decoder 200 already have been described above with reference to the decoder 40 of Figure 2. Therefore, only aspects of the decoder 200 of Figure 9 that are significantly different shall be described herein.

The FACCH detector 212 detects not only normal FACCH frames, but also detects FACCH/DTMF frames as well. When a FACCH/DTMF frame is received and detected, the controller 216 causes the DTMF generator 218 to produce an analog DTMF tone corresponding to the received FACCH/DTMF frame. The controller 216 also causes the second multiplexer 222 to select the DTMF tone produced by the DTMF generator 218 rather than the output of the first multiplexer 220.

The operation of the FSM 210 and the first multiplexer 220 will be appreciated from the following "Decoder Side FSM Stte Transition Table 'B'".

Decoder Side FSM State Transition Table "B"

State	Input	Next State	Output
Speech	FACCH/DTMF	DTMF	SILENCE
Speech	FACCH/SILENCE	SILENCE 1	SILENCE
Speech	Other	Speech	Speech
DTMF	Speech	SILENCE 1	SILENCE
DTMF	FACCH/SILENCE	SILENCE 1	SILENCE
DTMF	Other	DTMF	SILENCE
SILENCE 1	all	SILENCE 2	SILENCE
SILENCE 2	Speech	Speech	Speech
SILENCE 2	FACCH/DTMF	DTMF	SILENCE
SILENCE 2	Other	SILENCE 2	SILENCE

In this alternative embodiment, the FSM 210 does not directly control the selection of DTMF tones as the output. Rather, the controller 216 performs that role. More specifically, the controller 216 is programmed to interpret FACCH messages sent using $LAPD_m$ so as to identify an encoded DTMF tone, to instruct the DTMF generator 218 to produce the identified DTMF tone, and to instruct the second multiplexer 222 to select the produced tone.

The decoder embodiment of Figure 2 is preferable when there is likely to be a lower error rate, as for example, when there are relatively few cellular hand-offs. The decoder embodiment of Figure 9 is preferable when there is likely to be a higher bit error rate, as for example, when there are a relatively large number of hand-offs. The decoder 40 of Figure 2 produces higher fidelity because the duration of the DTMF tone is preserved. The decoder embodiment of Figure 2 also is generally somewhat less complex to implement than the decoder 200 of Figure 9, because it does not involve the controller 216 in processing DTMF-encoding frames using the $LAPD_m$ regeneration algorithm.

However, the decoder 40 of Figure 2 is more likely to miss bits at higher error rates because, if more than one frame in a row is missed then a DTMF-encoding of a DTMF tone may be lost because there is no retransmission. On the other hand, although the decoder embodiment of Figure 9 produces lower fidelity because the duration of the digits is fixed, it is less likely to miss bits at high bit error rates because the $LAPD_m$ regeneration algorithm is employed to ensure proper transmission of DTMF-encoding frames.

While particular embodiments of the invention have been described in detail, various modifications to the preferred embodiment can be made without departing from the spirit and scope of the invention. Thus, the invention is limited only by the appended claims.

WHAT IS CLAIMED IS:

1. A process for communicating dual tone multiple frequency (DTMF) tone information over wireless connections between encoders and decoders in a digital cellular telephone system comprising the steps of:

receiving in an encoder a DTMF tone;

identifying in the encoder the received DTMF tone;

designating in the encoder a DTMF-encoding frame that digitally encodes the identified DTMF tone; and

transmitting the designated DTMF-encoding frame from the encoder to a decoder by radio over a traffic channel of the digital cellular telephone system.

2. The process of claim 1 further including the steps of:

receiving in the encoder analog voice signals;

producing in the encoder speech coder signals from the received analog voice signals;

transmitting the speech coder signals from the encoder to the decoder over the traffic channel of the digital cellular system; and

blanking the speech coder signals during transmission of the DTMF-encoding frame.

3. The process of claim 2 wherein the encoder performs the step of blanking.

4. The process of claim 1 wherein the step of transmitting includes transmitting multiple of the designated DTMF-encoding frames.

5. The process of claim 1 wherein the step of transmitting includes transmitting at least two of the designated DTMF-encoding frames in sequence.

6. The process of claim 1 wherein the step of transmitting includes transmitting a plurality of the designated DTMF-encoding frames during a time interval when the DTMF tone is received in the encoder.

7. The process of claim 1 including the further steps of:

designating an end-of-DTMF tone frame which signifies that the received DTMF tone has stopped being received; and

transmitting the designated end-of-DTMF tone frame from the encoder to the decoder by radio over the traffic channel when the received DTMF tone has stopped being received in the encoder.

8. The process of claim 1, including the further steps of:

designating an end-of-DTMF tone frame which signifies that the received DTMF tone has stopped being received;

transmitting the designated end-of-DTMF tone frame from the encoder to the decoder by radio over the traffic channel when the received DTMF tone has stopped being received in the encoder; and

wherein the designated end-of-DTMF tone frame is a silence frame.

9. The process of claim 1 including the further steps of:

designating an end-of-DTMF tone frame to signify that the received DTMF tone has stopped being received; and

transmitting at least two of the designated end-of-DTMF tones frames from the encoder to the decoder by radio over the traffic channel when the received DTMF tone has stopped being received in the encoder.

10. The process of claim 1 including the further steps of:

designating an end-of-DTMF tone frame to signify that the received DTMF tone has stopped being received;

transmitting at least two of the designated end-of-DTMF tones frames from the encoder to the decoder by radio over the traffic channel when the received DTMF tone has stopped being received in the encoder; and

wherein the designated end-of-DTMF tone frame is a silence frame.

11. The process of claim 1, wherein

the step of transmitting includes transmitting at least two of the designated DTMF-encoding frames in sequence while the DTMF tone is received in the encoder; and

including the further steps of:

designating an end-of-DTMF tone frame to signify that the received DTMF tone has stopped being received; and

transmitting at least two of the designated end-of-DTMF tone frames from the encoder to the decoder by radio over the traffic channel when the received DTMF tone has stopped being received in the encoder

12. The process of claim 1,

wherein the step of transmitting includes transmitting at least two of the designated DTMF-encoding frames in sequence while the DTMF tone is received in the encoder; and

including the further steps of:

designating an end-of-DTMF tone frame to signify that the received DTMF tone has stopped being received;

transmitting at least two of the designated end-of-DTMF tone frames from the encoder to the decoder by radio over the traffic channel when the received DTMF tone has stopped being received in the encoder; and

wherein the designated end-of-DTMF tone frame is a silence frame.

13. The process of claim 1 wherein the designated DTMF-encoding frame is a Fast Associated Control Channel (FACCH) frame.

14. The process of claim 1,

wherein the designated DTMF-encoding frame is a Fast Associated Control Channel (FACCH) frame;

wherein the designated end-of DTMF tone frame is a FACCH frame;

wherein the step of transmitting includes transmitting at least two of the designated DTMF-encoding frames in sequence when the DTMF tone begins to be received and transmitting more of the produced DTMF-encoding frames while the DTMF tone is being received; and

including the further steps of:

designating an end-of-DTMF tone frame to signify that the received DTMF tone has stopped being received in the encoder; and

transmitting at least two of the designated end-of-DTMF tone frames from the encoder to the decoder by radio over the traffic channel when the received DTMF tone has stopped being received in the encoder.

15. The process of claim 2, wherein the step of blanking includes transmitting at least two of the designated DTMF-encoding frames in sequence over the traffic channel when the DTMF tone begins to be received in the encoder and transmitting a series of the designated DTMF-encoding frames while the DTMF tone is being received in the encoder.

16. The process of claim 1 including the further step of applying forward error correction to the DTMF-encoding frames prior to said step of transmitting.

17. The process of claim 1,

wherein said step of transmitting the designated DTMF-encoding frame from the encoder to the decoder includes transmitting multiple of the DTMF-encoding frames while the DTMF tone is received in the encoder; and including the further steps of:

identifying in the decoder the DTMF tone represented by the DTMF-encoding frame; and

generating in the decoder the identified DTMF tone while the encoder transmits the multiple DTMF-encoding frames.

18. The process of claim 1,

wherein said step of transmitting the designated DTMF-encoding frame from the encoder to a decoder includes transmitting multiple of the DTMF-encoding frames while the DTMF tone is received in the encoder; and including the further steps of:

identifying in the decoder the DTMF tone represented by the DTMF-encoding frame; and

generating in the decoder the identified DTMF tone while the encoder transmits the multiple DTMF-encoding frames;

receiving in the encoder, after said step of receiving in the decoder the DTMF tone, analog voice signals in the encoder the DTMF tone;

producing in the encoder speech coder signals from the received analog voice signals;

transmitting the speech coder frames from the encoder to the decoder over the traffic channel of the digital cellular system;

designating an end-of-DTMF tone frame to signify that the received DTMF tone has stopped being received in the encoder;

identifying in the encoder the end of the received DTMF tone;

transmitting at least two of the designated end-of-DTMF tone frames from the encoder to the decoder by radio over the traffic channel when the end of the DTMF tone has been identified; and

following generation of the identified DTMF tone by the decoder, providing analog speech signals as decoder output only after the decoder receives the at least two end-of-DTMF tone frames from the encoder or after the decoder receives at least two speech coder frames from the decoder.

19. The process of claim 18 further including the step of:

following generation of the identified DTMF tone by the decoder, providing analog speech signals as decoder output only after the decoder provides silence as an output in response to the at least two end-of-DTMF tone frames or in response to the at least two speech coder frames.

20. The process of claim 1 including the further steps of:

receiving the DTMF-encoding frame in the decoder;

identifying in the decoder the DTMF tone represented by the DTMF-encoding frame; and

applying a retransmission algorithm to the DTMF-encoding frame in the decoder;

after said step of applying the retransmission algorithm, identifying in the decoder the DTMF tone represented by the DTMF-encoding frame; and

generating in the decoder the identified DTMF tone.

21. The process of claim 20 further including the steps of:

receiving in the encoder analog voice signals;

producing in the encoder speech coder signals from the received analog voice signals;

transmitting the speech coder signals from the encoder to the decoder over the traffic channel of the digital cellular system; and

blanking the speech coder signals during transmission of the DTMF-encoding frame.

22. The process of claim 21 wherein the decoder performs the step of blanking.

23. An encoder for use in a digital cellular telephone system comprising:

an interface to an analog telephone;

a radio tuneable to a traffic channel of the digital cellular telephone system;

means for producing speech coder signals in response to analog voice signals received on the telephone interface;

a DTMF detector;

means for producing a DTMF-encoding frame in response to DTMF tone received on the telephone interface; and

means for alternatively providing the speech coder signals or the DTMF-encoding frame to the radio tuned to the traffic channel.

24. The encoder of claim 23 further including:

means for blanking the speech coder signals during radio transmission of the DTMF-encoding frame over the traffic channel.

25. The encoder of claim 23 wherein,

said means for producing DTMF-encoding frames produces at least two DTMF-encoding frames in response to each DTMF tone received on the telephone interface; and

said means for alternatively providing provides the at least two DTMF-encoding frames to the radio tuned to the traffic channel.

26. The encoder of claim 23 wherein,

said means for producing DTMF-encoding frames produces multiple DTMF-encoding frames during an interval when the DTMF tone is received on the telephone interface; and

said means for alternatively providing provides the multiple DTMF-encoding frames to the radio tuned to the traffic channel.

27. The encoder of claim 23 further including:

means for producing an end-of-DTMF tone frame which signifies that the received DTMF tone has stopped being received;

wherein said means for alternatively providing alternately provides the speech coder signals or the DTMF-encoding frame or the end-of-DTMF tone frame to the radio tuned to the traffic channel.

28. The encoder of claim 23,

wherein said means for producing DTMF-encoding frames produces multiple DTMF-encoding frames during an interval when the DTMF tone is received on the telephone interface; and

further including:

means for producing an end-of-DTMF tone frame which signifies that the received DTMF tone has stopped being received;

wherein said means for alternatively providing alternately provides the speech coder signals or multiple DTMF-encoding frames or the end-of-DTMF tone frame to the radio tuned to the traffic channel such that the DTMF-encoding frames are provided while a DTMF tone is received on the telephone interface and the end-of-DTMF tone frame is provided when the DTMF tone has stopped being received on the telephone interface.

29. The encoder of claim 23,

wherein said means for producing the DTMF-encoding frame produces a Fast Associated Control Channel (FACCH) frame.

30. The encoder of claim 23 further including:

means for producing an end-of-DTMF tone Fast Associated Control Channel (FACCH) frame which signifies that the received DTMF tone has stopped being received;

wherein said means for alternatively providing alternately provides the speech coder signals or the DTMF-encoding frame or the end-of-DTMF tone frame to the radio tuned to the traffic channel.

31. The encoder of claim 30 wherein the end-of-DTMF tone FACCH frame is a silence frame.

32. The encoder of claim 23 further including:

means for applying forward error correction to speech coder signals and to DTMF-encoding frames provided to the radio by said means for providing.

33. A decoder for use in a digital cellular telephone system comprising:

an interface to an analog telephone;

a radio tuneable to a traffic channel of the digital cellular telephone system;

means for producing analog speech signals in response to speech coder signals received on the radio tuned to the traffic channel;

means for producing a DTMF tone in response to a DTMF-encoding frame received on the radio tuned to the traffic channel; and

means for alternatively providing the analog speech signals or the DTMF tone on the telephone interface.

34. The decoder of claim 33, further including:

means for identifying an end-of-DTMF tone frame;

wherein said means for alternatively providing the analog speech signals or the DTMF tone on the telephone interface provides the analog speech signals on the telephone interface following generation of a DTMF tone only, after identification of an end-of-DTMF tone frame following generation of the DTMF tone, or after producing analog speech signals in response to at least two speech coder signals following generation of the DTMF tone.

35. The system of claim 33,

wherein the decoder further includes,

retransmission means for ensuring that encoded DTMF digits are received in the proper sequence and for ensuring that DTMF digits are not missed.

36. The system of claim 33,

wherein the decoder further includes,

retransmission means for ensuring that encoded DTMF digits are received in the proper sequence and for ensuring that DTMF digits are not missed before said means for producing a DTMF tone produces such DTMF tone in response to the received DTMF-encoding frame.

37. A digital cellular telephone system comprising:

an encoder which includes,

an interface to an analog telephone;

a radio tuneable to a traffic channel of the digital cellular telephone system;

means for producing speech coder signals in response to analog voice signals received on the telephone interface;

a DTMF detector;

means for producing a DTMF-encoding frame in response to DTMF tone received on the telephone interface; and

means for alternatively providing the speech coder signals or the DTMF-encoding frame to the radio tuned to the traffic channel; and

a decoder which includes,

an interface to an analog telephone;

a radio tuneable to a traffic channel of the digital cellular telephone system;

means for producing analog speech signals in response to speech coder signals received on the radio tuned to the traffic channel;

means for producing a DTMF tone in response to a DTMF-encoding frame received on the radio tuned to the traffic channel; and

means for alternatively providing the analog speech signals or the DTMF tone on the telephone interface.

38. The system of claim 37,

wherein the encoder further includes,

means for producing an end-of-DTMF tone frame which signifies that the received DTMF tone has stopped being received;

wherein said means for alternatively providing alternately provides the speech coder signals or multiple DTMF-encoding frames or the end-of-DTMF tone frame to the radio tuned to the traffic channel such that the DTMF-encoding frames are provided while a DTMF tone is received on the telephone interface and the end-of-DTMF tone frame is provded when the DTMF tone has stopped being received on the telephone interface; and

wherein the decoder further includes,

means for identifying the end-of-DTMF tone frame;

wherein said means for alternatively providing the analog speech signals or the DTMF tone on the telephone interface provides the analog speech signals on the telephone inteface following generation of a DTMF tone only, after identification of an end-of-DTMF tone frame following generation of the DTMF tone, or after producing analog speech signals in response to at least two speech coder signals following generation of the DTMF tone.

39. The system of claim 37,

wherein the decoder further includes,

retransmission means for ensuring that encoded DTMF digits are received in the proper sequence and for ensuring that DTMF digits are not missed.

40. The system of claim 37,

wherein the decoder further includes,

retransmission means for ensuring that encoded DTMF digits are received in the proper sequence and for ensuring that DTMF digits are not missed before said means for producing a DTMF tone produces such DTMF tone in response to the received DTMF-encoding frame.

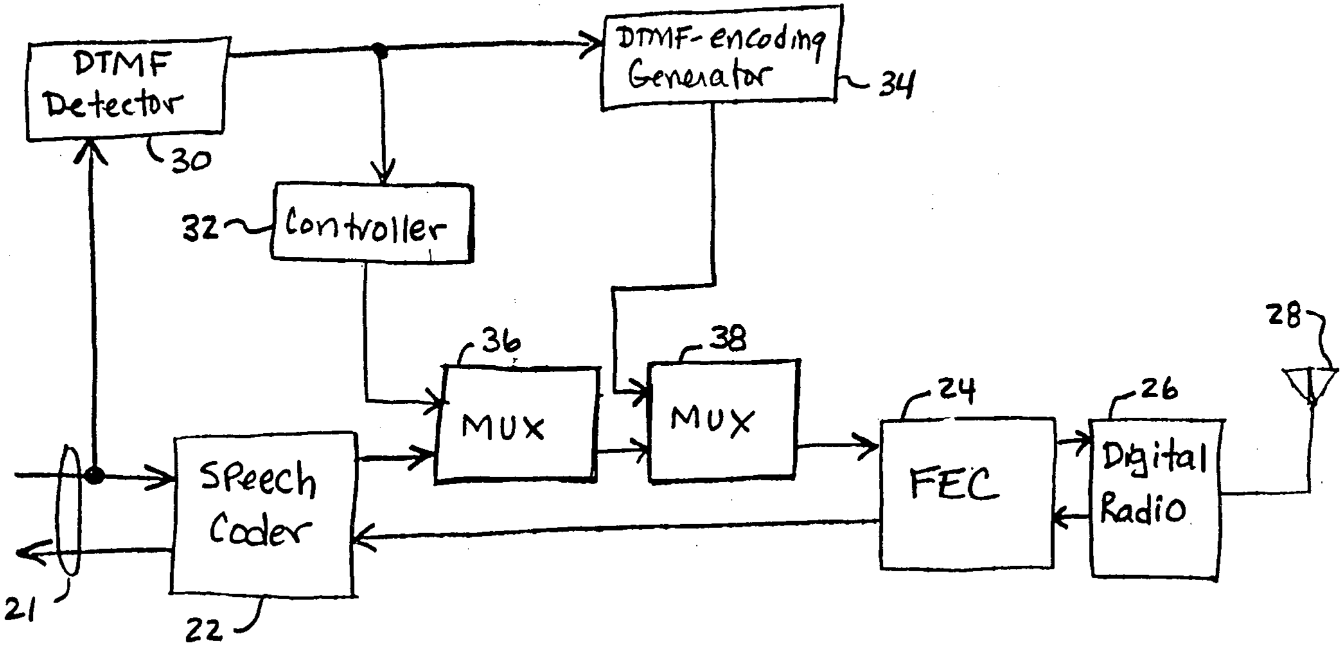
DTMF Detector
30
DTMF-encoding Generator
34
32
Controller
36
38
28
MUX
MUX
24
26
FEC
Digital Radio
Speech Coder
21
22

Figure 1

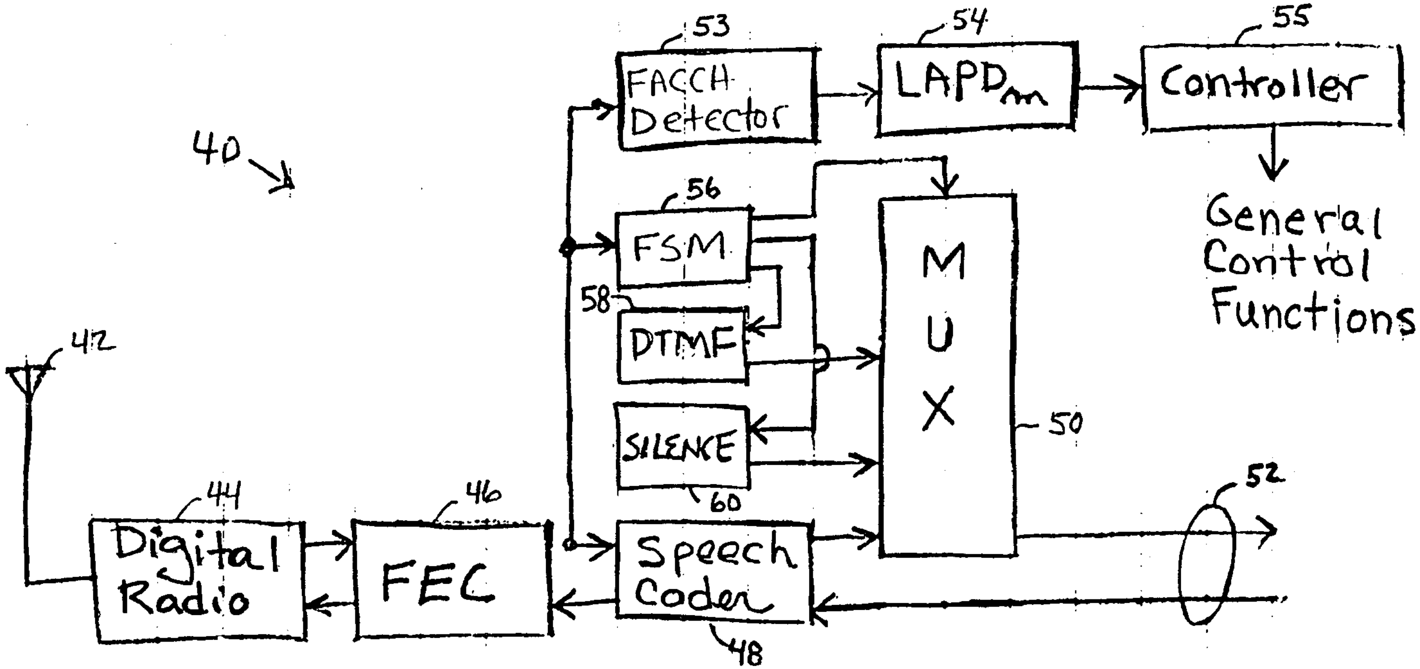
40
42
44
Digital Radio
46
FEC
48
Speech Coder
53
FACCH Detector
54
LAPDm
55
Controller
General Control Functions
56
FSM
58
DTMF
SILENCE
60
MUX
50
52

Figure 2

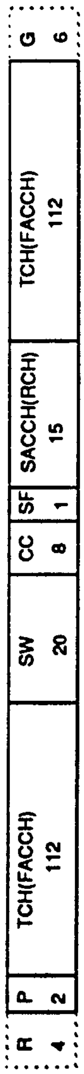

Figure 3A

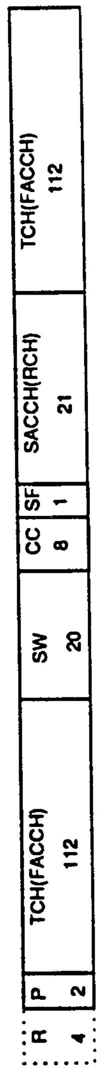

Figure 3B

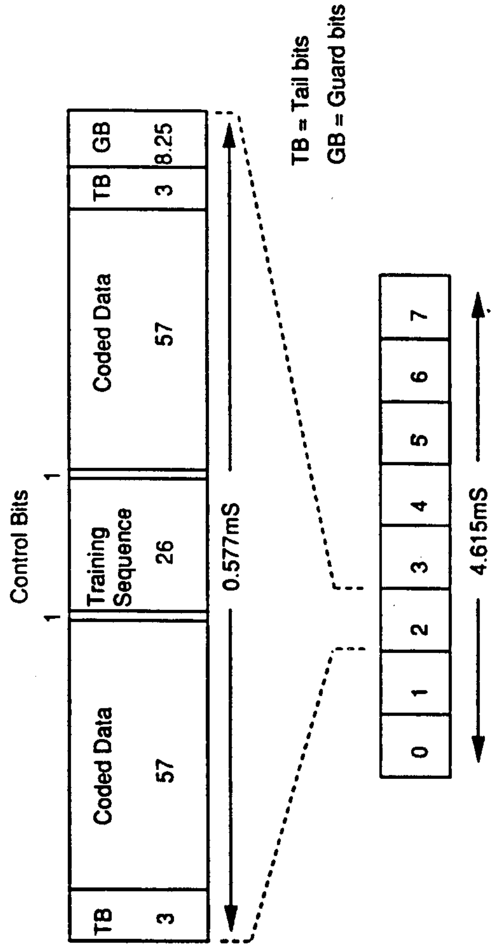
Control Bits
1
1
TB
3
Coded Data
57
Training Sequence
26
Coded Data
57
TB
3
GB
8.25
0.577mS
TB = Tail bits
GB = Guard bits
0
1
2
3
4
5
6
7
4.615mS

Figure 4

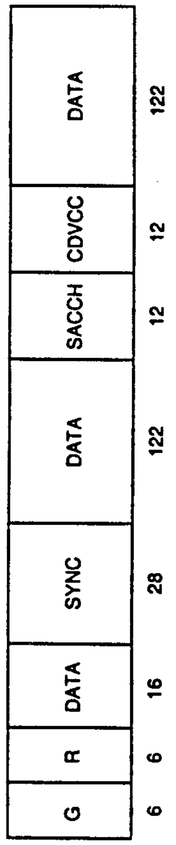

Figure 5A

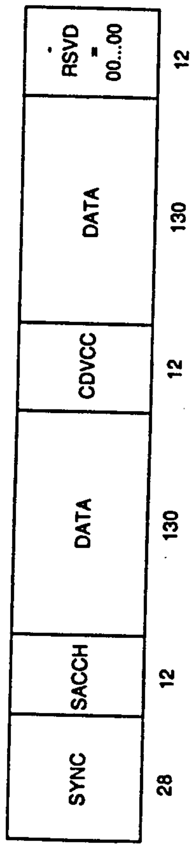

Figure 5B

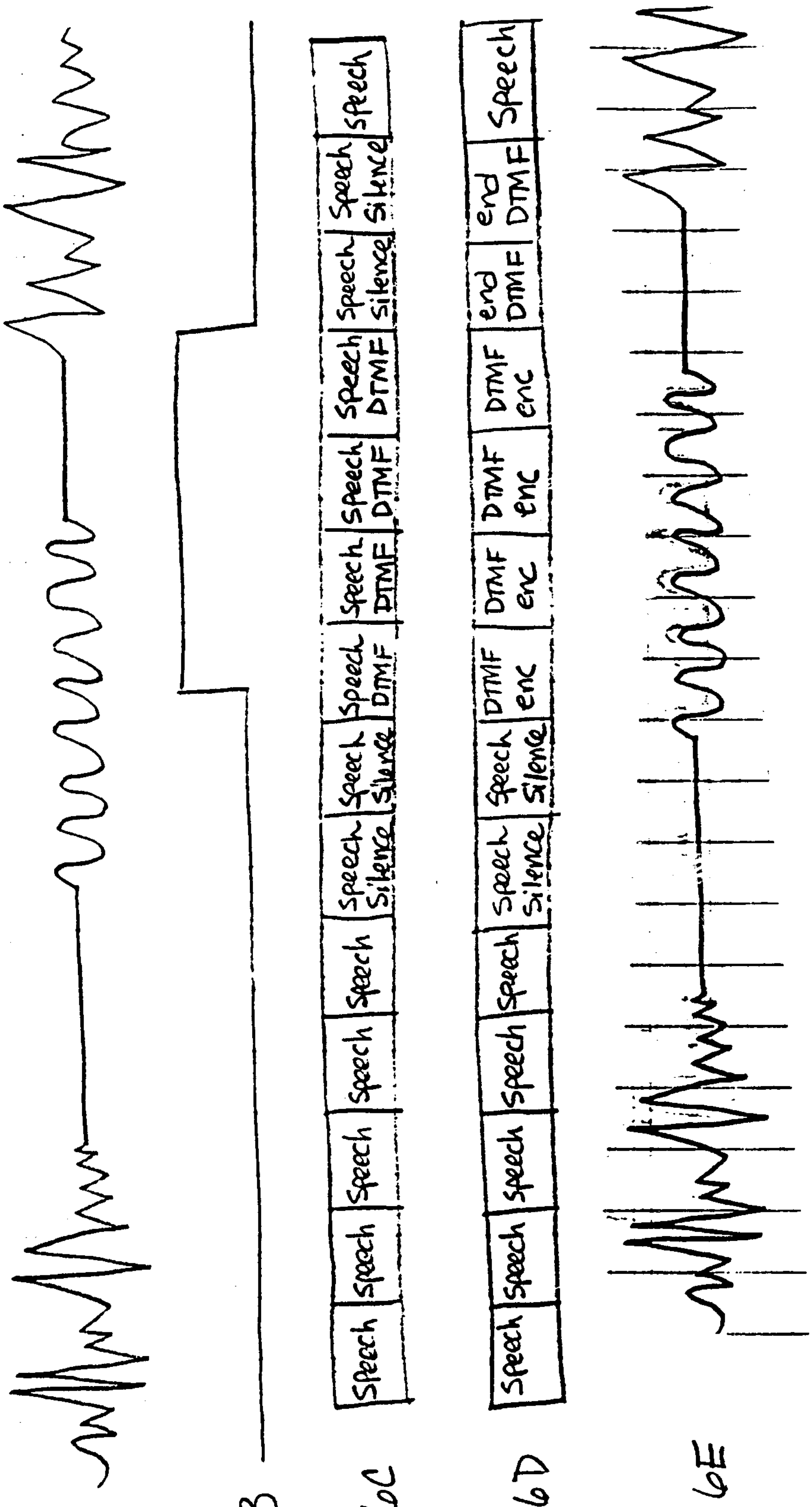

Figure 6A

Figure 6B

Figure 6C

Figure 6D

Figure 6E

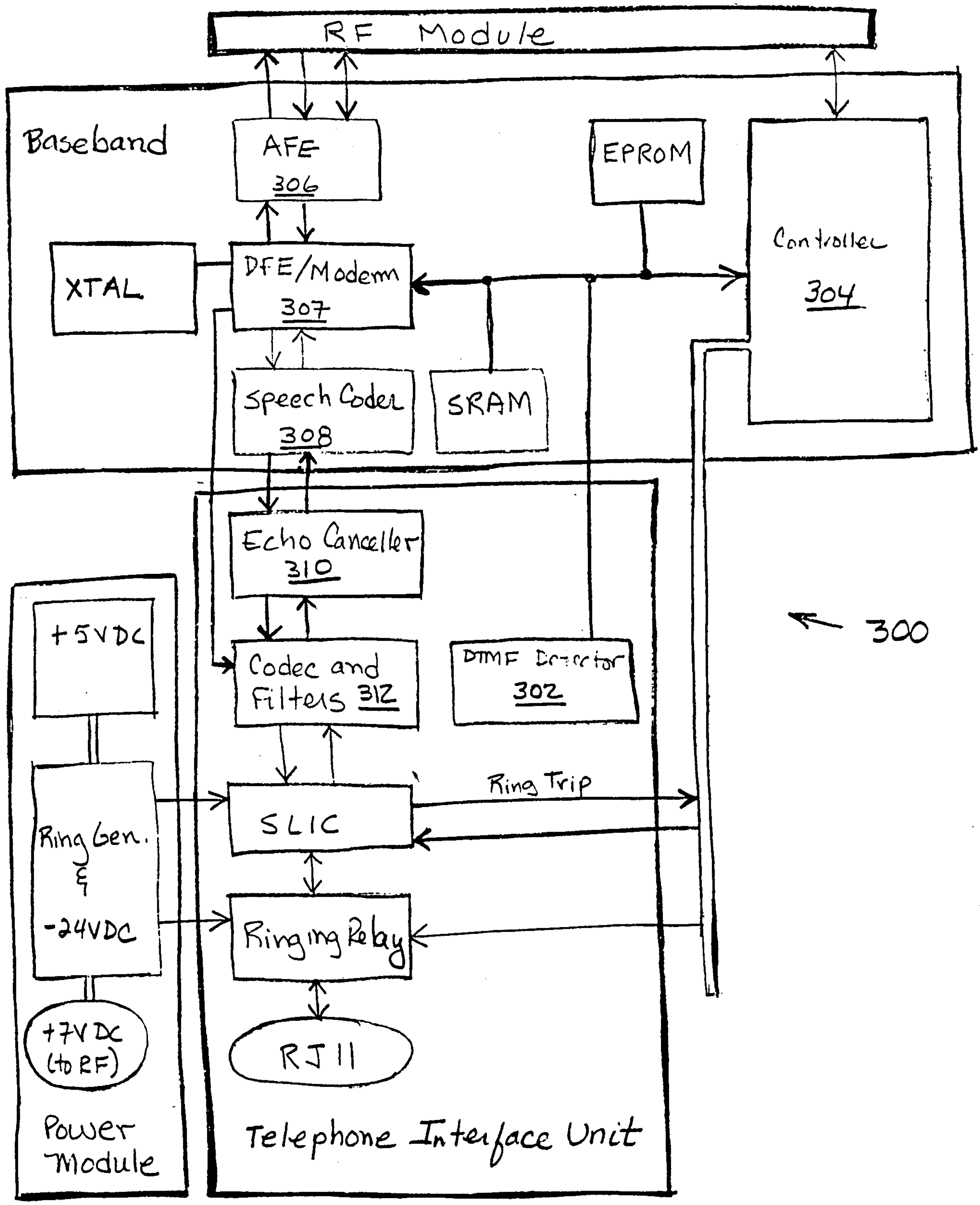
RF Module
Baseband
AFE
306
EPROM
XTAL
DFE/Modem
307
Controller
304
Speech Coder
308
SRAM
Echo Canceller
310
+5VDC
Codec and Filters 312
DTMF Detector
302
300
Ring Trip
SLIC
Ring Gen. & -24VDC
Ringing Relay
+7VDC (to RF)
RJ11
Power Module
Telephone Interface Unit

Figure 7

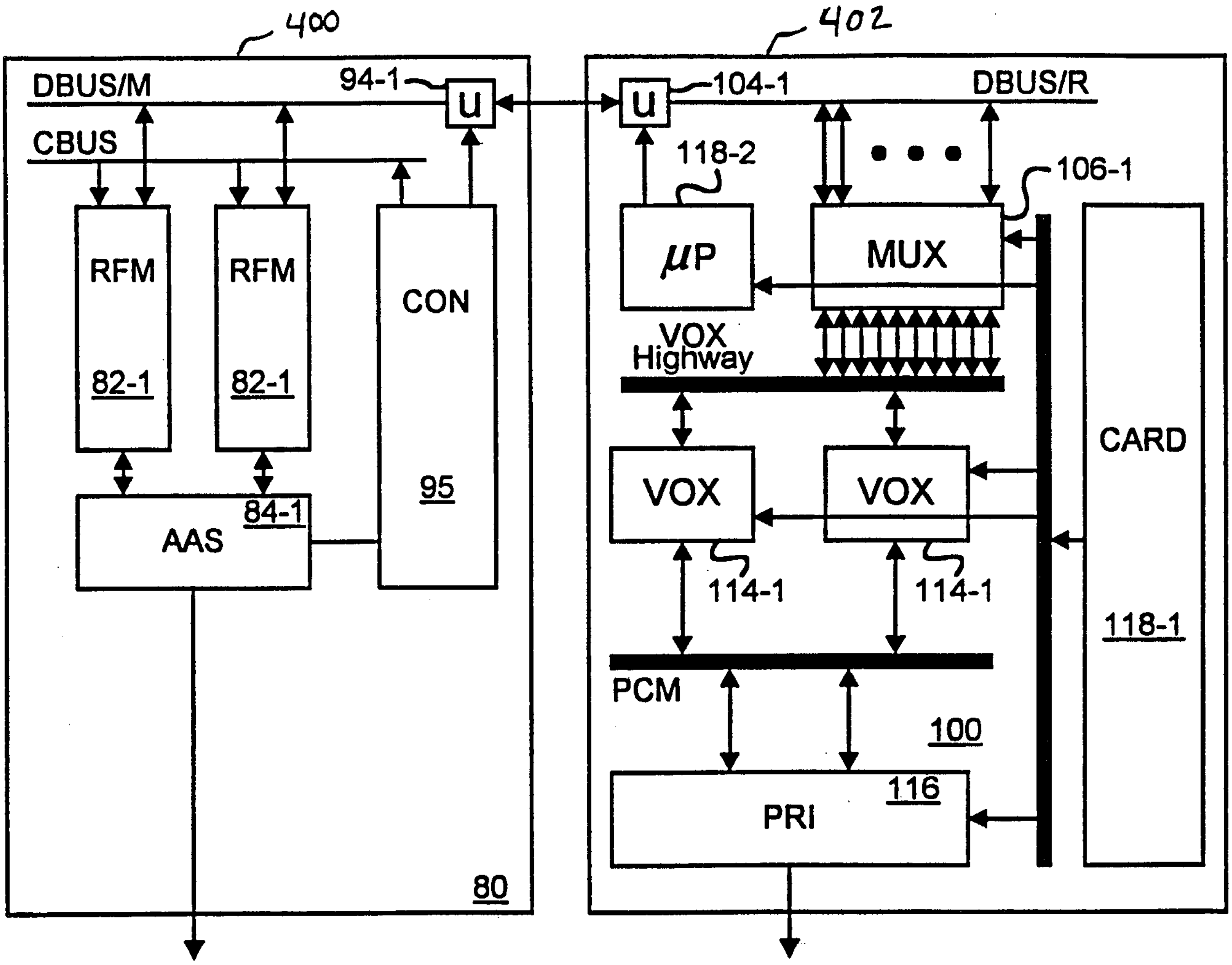
400
402
DBUS/M
94-1
u
CBUS
RFM
82-1
RFM
82-1
CON
95
AAS
84-1
80
u
104-1
DBUS/R
118-2
106-1
μP
MUX
VOX
Highway
CARD
118-1
VOX
VOX
114-1
114-1
PCM
100
PRI
116

Figure 8

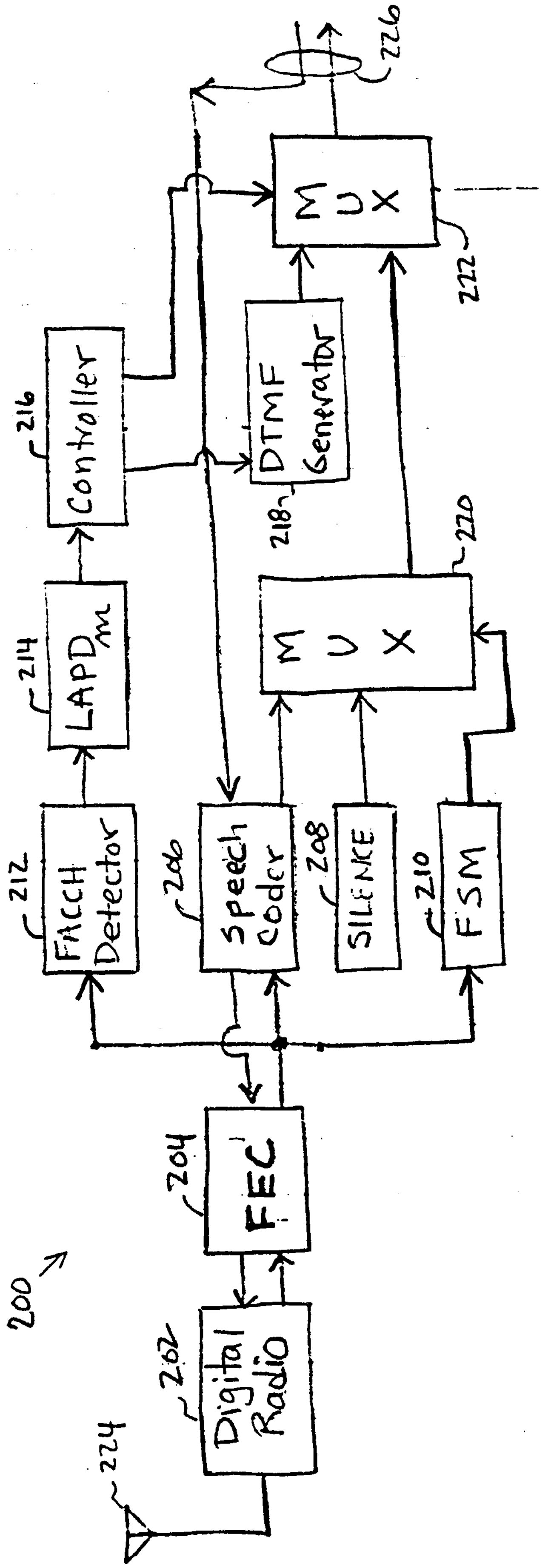
200
224
202
Digital Radio
204
FEC
212
FACCH Detector
206
Speech Coder
208
SILENCE
210
FSM
214
LAPDm
216
Controller
218
DTMF Generator
220
MUX
222
MUX
226

Figure 9