Digital Cellular Phone: A Functional Analysis

Application Report

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Introduction

This document presents the functional components of a dual-mode cellular phone as specified by the CTIA IS-54 standard. For each functional component, the relevant algorithm, its data structures, if any, and implementation details are given.

A Functional View of a Dual-Mode Cellular Phone

As shown in Figure 1, a dual-mode cellular phone consists of the following:

- Transmitter
- Receiver
- Coordinator
- Antenna assembly
- Control panel

A dual-mode phone is capable of operating in an analog-only cell or a dual-mode cell. Both the transmitter and the receiver support both analog FM and digital time division multiple access (TDMA) schemes. Digital transmission is preferred, so when a cellular system has digital capability, the mobile unit is assigned a digital channel first. If no digital channels are available, the cellular system will assign an analog channel.

The transmitter converts the audio signal to a radio frequency (RF), and the receiver converts an RF signal to an audio signal. The antenna focuses and converts RF energy for reception and transmission into free space. The control panel serves as an input/output mechanism for the end user; it supports a keypad, a display, a microphone, and a speaker. The coordinator synchronizes the transmission and receive functions of the mobile unit.

Figure 1. Functional Components of a Dual-Mode (IS-54) Cellular Phone

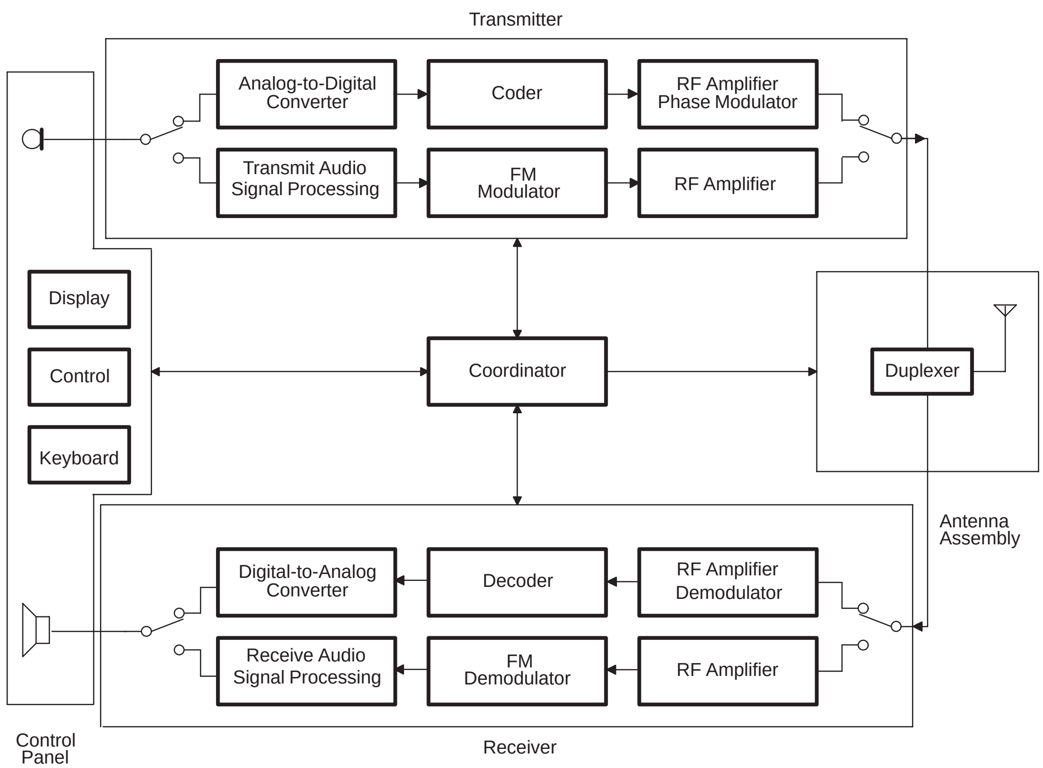
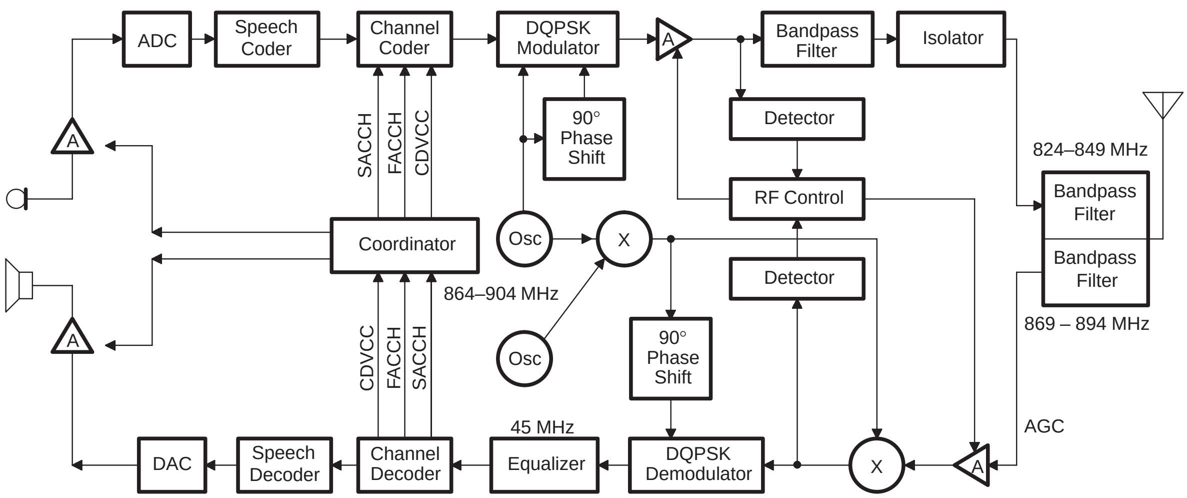


Figure 2 shows the functional components of the *digital portion* of a dual-mode cellular phone.

Figure 2. Functional Blocks of the Digital Portion of a Dual-Mode Phone



CDVCC = coded digital verification color code
DQPSK = differential quaternary phase-shift keying
FACCH = fast associated control channel
SACCH = slow associated control channel

Transmitter

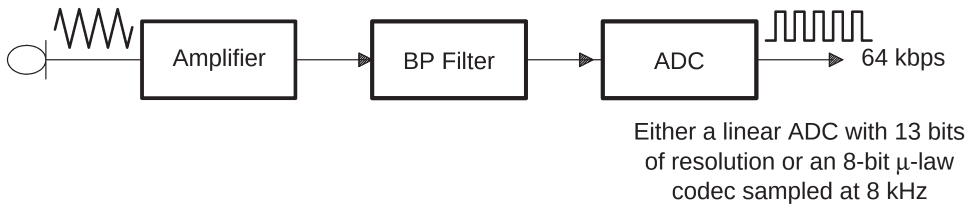
The transmitter converts low-level audio signals from the microphone to digitally coded RF signals by audio processing, digital signal processing, modulation, and RF amplification. The transmitter converts 64-kbps pulse code modulation (PCM) data to a lower data rate, multiplexes control information, error-protects the data, and then passes the data stream to the RF section for modulation, amplification, and transmission. The coordinator inserts system control messages.

Transmit Front-End Processing

Speech signals from the microphone are first amplified, passed through an antiliasing filter, and sampled at a rate of 8 kHz to create a digitized μ-law 64-kbps bit stream. Typically, no pre-emphasis is applied. Figure 3 shows the functional blocks of the front-end analog section. The standard does not propose any specific echo canceler; however, it recommends implementing one. The front-end processing includes the following:

- An amplifier. The gain is specified to produce an average signal energy, during a frame, which is 18 dB down from full scale.
- A bandpass filter to avoid antialiasing.
- An analog-to-digital converter. The standard recommends that you either directly convert the analog signal to a uniform PCM format with a minimum resolution of 13 bits or convert the analog signal to an 8-bit μ-law codec sample.

Figure 3. Front-End Analog Section Converts Audio to a 64-kbps Data Stream

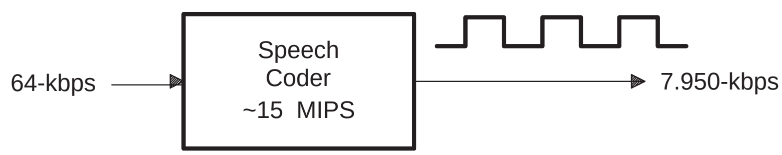


Speech Coder

The speech coder further reduces the data rate by compressing the 64-kbps data stream input to create a 7.950-kbps data stream. The IS-54 standard accepts a full-rate speech coder called *vector sum excited linear prediction* (VSELP). This algorithm belongs to a class of speech coders known as *code excited linear predictive coders* (CELP). This class uses code books to vector quantize the excitation (residual) signal. VSELP is a variation on CELP.

The incoming 64 kbps of data are grouped into frames at a frame rate of 50 frames per second. Hence, each frame contains 160 samples and represents a duration of 20 ms. Each frame is coded into 159 bits. Hence, the rate of the conversions is $50 \times 159 = 7950$ bps, as shown in Figure 4.

Figure 4. Full-Rate Speech Coder (VSELP) Reduces a 64-kbps Data Stream to an 8-kbps Data Stream



The speech decoder utilizes two separate code books. Each code book has an independent gain. The two code-book excitations are each multiplied by their corresponding gains and summed to create a combined code-book excitation. The basic parameters are shown in Table 1.

Table 1. Basic Parameters of a VSELP Speech Coder

Parameter	Notation	Specification
Sampling rate	s	8 kHz
Frame length	Nf	160 samples (20 ms)
Subframe length	N	40 samples (5 ms)
Short-term predictor order	Np	10
Number of taps for long-term predictor	N_L	1
Number of bits in code word 1 (number of basis vectors)	M1	7 bits
Number of bits in code word 2 (number of basis vectors)	M2	7 bits

NOTE: Within a frame, the 159 bits are allocated as shown in Table 2; detailed bit allocations are shown in Table 3.

Table 2. Bit Allocations Within a Frame of Speech

Parameter	Bits Allocated
Short-term filter coefficients	38
Frame energy, R0	5
Lag, L	28
Code words, I, H	56
Gains beta, gamma1, gamma2	32

Table 3. Detailed Bit Allocations of Parameters Within a Frame

Parameter	Parameter Name	Bits Allocated
Frame energy	R0	5
1st reflection coefficient	LPC1	6
2nd reflection coefficient	LPC2	5
3rd reflection coefficient	LPC3	5
4th reflection coefficient	LPC4	4
5th reflection coefficient	LPC5	4
6th reflection coefficient	LPC6	3
7th reflection coefficient	LPC7	3
8th reflection coefficient	LPC8	3
9th reflection coefficient	LPC9	3
10th reflection coefficient	LPC10	2
Lag for first subframe	LAG_1	7
Lag for second subframe	LAG_2	7
Lag for third subframe	LAG_3	7
Lag for fourth subframe	LAG_4	7
1st code book, I, for first subframe	CODE1_1	7
1st code book, I, for second subframe	CODE1_2	7
1st code book, I, for third subframe	CODE1_3	7
2nd code book, H, for first subframe	CODE2_1	7
2nd code book, H, for second subframe	CODE2_2	7
2nd code book, H, for third subframe	CODE2_3	7
2nd code book, H, for fourth subframe	CODE2_4	7
{GS, P0, P1} code for first subframe	GSP0_1	8
{GS, P0, P1} code for second subframe	GSP0_2	8
{GS, P0, P1} code for third subframe	GSP0_3	8
{GS, P0, P1} code for fourth subframe	GSP0_4	8

Channel Coder

The main function of the channel coder is to protect the data stream against the noise and fading that are inherent to a radio channel. The coder accomplishes this by adding extra or redundant bits. The greater the number of redundant bits, the higher the immunity to interference and the lower the bit-error rate. The tradeoff is an increased data rate.

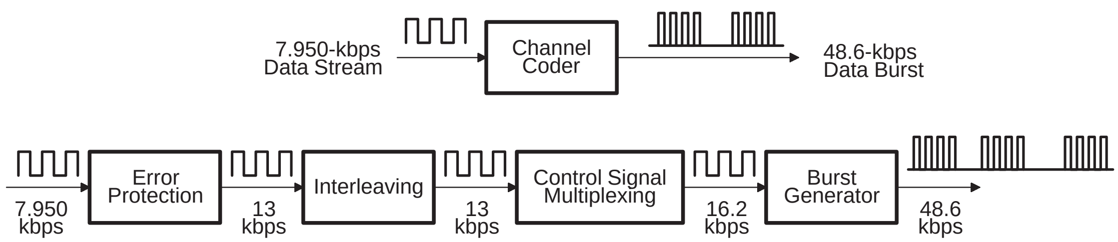
The channel coder protects the data stream in four stages:

1. Convolutional coding
2. Cyclic redundancy check (CRC) generation
3. Interleaving
4. Burst generation

The first two are *mathematical* operations, whereas the last two are *heuristic* approaches. The receiver performs an inverse operation to determine whether errors have occurred during propagation. In radio propagation, it has been found that the fading occurs at localized instances of time and space. As a result, interleaving spreads the information of the data stream across two frames, because it is unlikely that a clustered bit error would occur in successive frames. Finally, data is propagated in bursts.

Between interleaving and burst generation, the channel coder multiplexes control information. Figure 5 shows the functional components of a channel coder.

Figure 5. A Channel Coder and Its Functional Components With Associated Data Rates



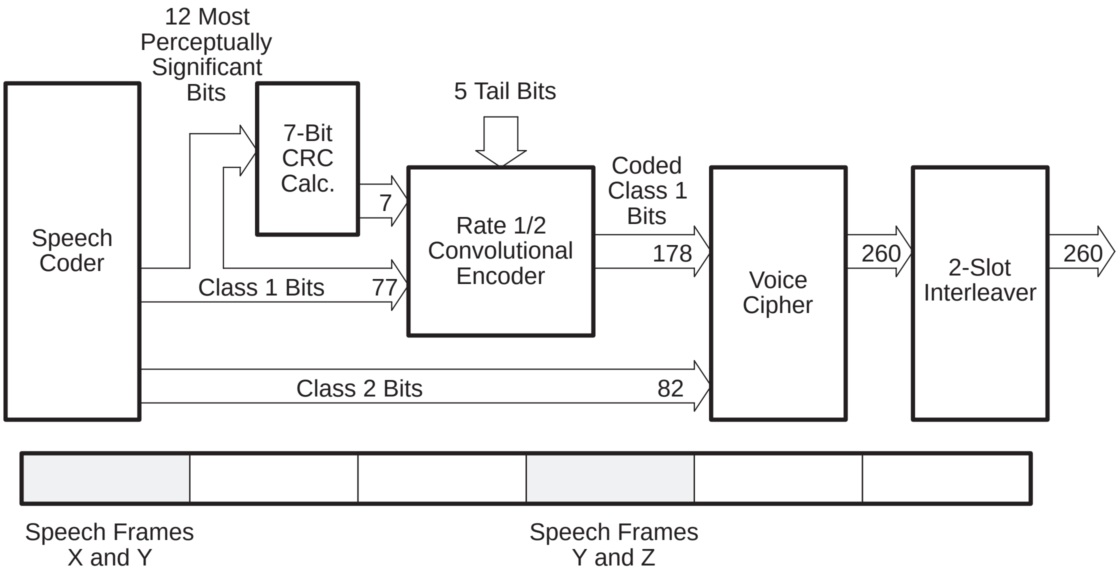
Convolutional Coding

Convolutional coding provides error-correction capability by adding redundancy to the transmitted sequence. Convolutional encoding is implemented by linear feed-forward shift registers.

A convolutional coder is described by the rate at which data enters the coder and the rate at which data leaves the coder. For example, a rate-1/2 convolutional coder implies that for every 1 bit of data entering the coder, 2 bits leave the coder. The smaller the ratio, the greater the redundancy. This improves the error-protection capability.

To reduce the bit rate, not all of the 159 bits in a frame are error-protected. Only 77 of these bits, called class 1 bits, are error-protected. The remaining 82 bits, called class 2 bits, are not error-protected. This is shown in Figure 6.

Figure 6. Error Protection via Convolutional Coding and CRC Computation



Cyclic Redundancy Check

Of the 77 bits that are error-protected, it has been found that only 12 are perceptually significant. Hence these are protected by using a 7-bit cyclic redundancy computation before they are input to the convolutional coder. A 7-bit CRC is computed by dividing the data by a specified constant and transmitting the remainder with the data. The receiver detects errors by comparing the received remainder with what it has calculated.

The following generator polynomial is used for the CRC:

$$gCRC(X) = 1 + X + X^2 + X^4 + X^5 + X^7 \quad (1)$$

The parity polynomial, b(X), is the remainder of the division of the input polynomial by the generator polynomial as shown below:

$$a(X)*X^7 / gCRC(X) = q(X) + b(X)/gCRC(X) \quad (2)$$

where q(X) is the quotient of the division and b(x) is the remainder. The quotient is discarded, and only the parity bits identified in b(X) are encoded for transmission. To facilitate the convolutional coder, these parity bits are placed into the array of class 1 bits.

Figure 7. Error Protection Adds 101 Extra Bits per Speech Frame

In short, as shown in Figure 7, error protection adds 101 bits every 20 ms, or an additional 5050 bps.

Interleaving

As explained earlier, data from each frame is now divided and spread across two transmit slots. This is done because fading might destroy a frame, but it is unlikely that it will destroy two frames in succession. As a result, not all bits from a speech frame are lost by one bad slot. Figure 8 shows how the data is interleaved when x, y, and z are three speech frames in succession.

Figure 8. Interleaving Adjacent Frames for Error Protection

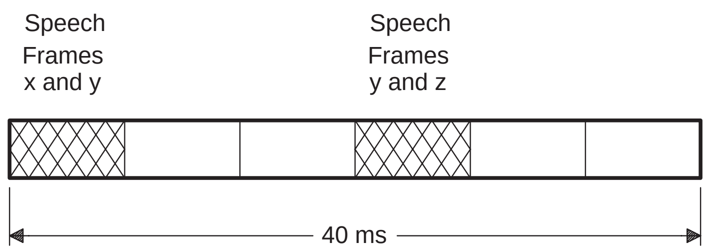


Table 4 shows how the data is interleaved when y is the current frame and x is the previous frame. Note that the speech data is entered into the interleaving array by columns.

Table 4. Interleaving of Two Adjacent Speech Frames, x and y

x0	x26	x52	x78	x104	x130	x156	x182	x208	x234
y1	y27	y53	y79	y105	y131	y157	y183	y209	y235
x2	x28	x54	x80	x106	x132	x158	x184	x210	x236
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
x12	x38	x64	x90	x116	x142	x168	x194	x220	x246
y13	y39	y65	y91	y117	y143	y169	y195	y221	y247
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
x24	x50	x76	x102	x128	x154	x180	x206	x232	x258
y25	y51	y77	y103	y129	y155	y181	y207	y233	y259

The 159 bits from a speech frame are classified as class 1 and class 2 bits; data is placed into the interleaving array in such a way that class 2 bits are intermixed with class 1 bits. Class 2 bits are sequentially placed into the array and occupy the following numbered locations:

0,	26,	52,	78
93	through		129
130,	156,	182,	208
223	through		259

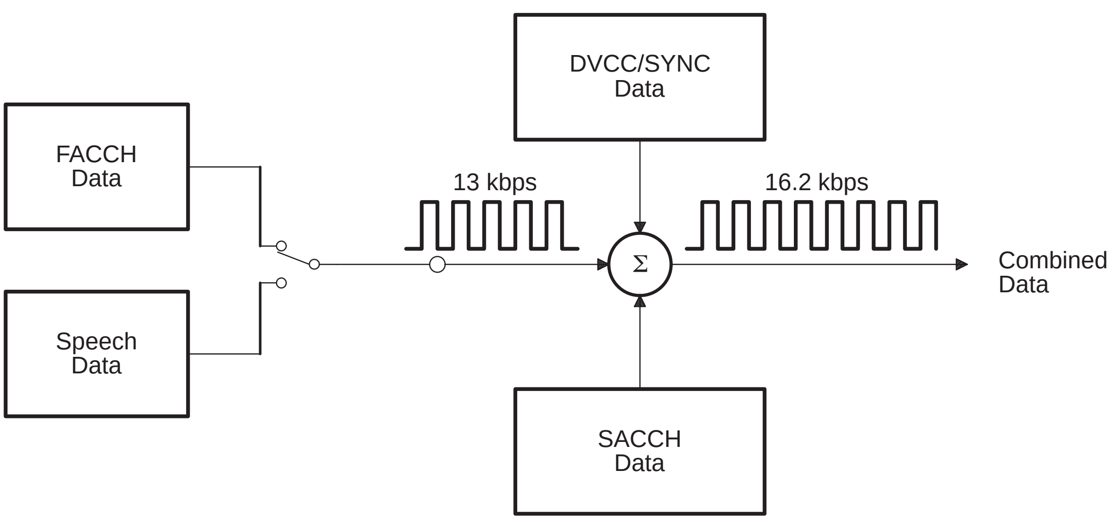
Control Signal Multiplexing

Control signal information is added to the interleaved data. Control information includes

- Slow associated control channel (SACCH)
- Fast associated control channel (FACCH)
- Digital verification color code (DVCC)
- Synchronization word (SYNC)

Figure 9 shows how all this control information is multiplexed.

Figure 9. Control-Signal Multiplexing



Slow associated control channel (SACCH) is a signaling channel in parallel with the speech path used for the transmission of control and supervisory messages between the base station and the mobile unit. SACCH messages are continuously mixed with the channel data; 12 bits are allocated for SACCH.

Fast associated control channel (FACCH) is a signaling channel for the transmission of control and supervisory messages between the base station and the mobile unit. FACCH messages are not mixed with the user information bits; they replace the user information block whenever necessary.

Digital verification color code (DVCC) is an 8-bit code that is sent by the base station to the mobile unit and is used to generate coded digital verification color code (CDVCC). CDVCC is a 12-bit field that includes the 8-bit DVCC; CDVCC is sent in each slot from the base station to the mobile unit and vice versa. The CDVCC is used by the receiver to distinguish the current traffic channel from traffic cochannels.

Synchronization word (SYNC) is a 14-symbol field that is used for slot synchronization, equalizer training, and time slot identification.

Mobile Assisted Handoff

Mobile Assisted Handoff (MAHO) is a new feature of IS-54. The base station can command the mobile unit to perform signal quality measurements on the current forward channel and any other 12 forward channels. The mobile unit can measure two quantities:

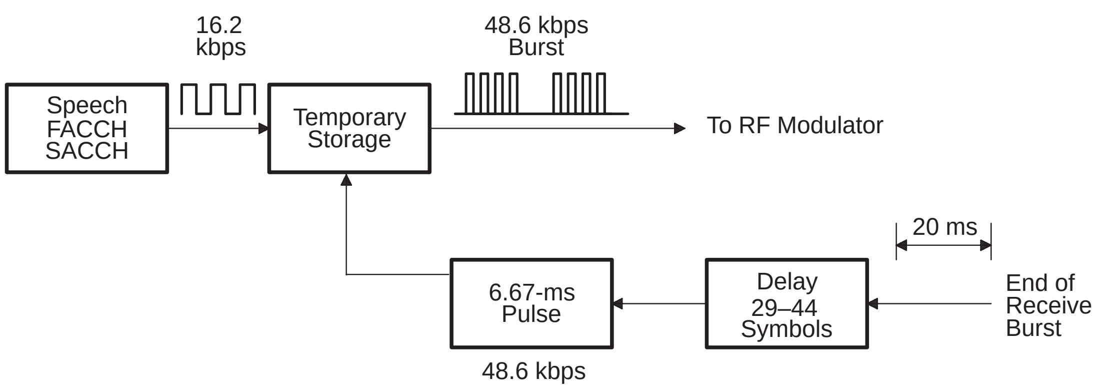
1. Received signal strength indicator (RSSI), which is a measure of the signal strength expressed in dB.
2. Bit error rate (BER), which is an estimate of the bit error information obtained by measuring the correctness of the data stream at the input to the mobile unit's channel decoder.

These channel quality measurements (RSSI and BER) are sent to the base station to assist it in handoff. This reduces the overhead on the base station. RSSI and BER are usually sent via SACCH, although they could be sent via FACCH during discontinuous transmission (DTX). DTX is a mode of operation in which a mobile unit transmitter autonomously switches between two transmitter power levels while the mobile unit is in the conversation state on an analog voice channel or a digital traffic channel.

Burst Generator

After the data has been compressed and error-protected, the bit stream is compressed (in time only) into a burst format. Burst timing offsets may be applied to facilitate dynamic time alignment. Figure 10 shows how the data is compressed and time-aligned to allow the data to be sent using one-third of the 48.6-kbps channel.

Figure 10. Burst Generator



Transmitter π/4 DQPSK Modulator and RF Amplifier

The 48.6-kbps data is now input to a *differential quaternary phase-shift keying* (DQPSK) modulator. This phase modulator groups two bits at a time to create a symbol. This results in four levels of modulation, as shown in Figure 11. Hence, the name quaternary. The term *differential* is used because symbols are transmitted as relative phase changes, rather than absolute phase values.

Figure 11. A 4-Level Modulator Groups Two Bits to Form a Symbol

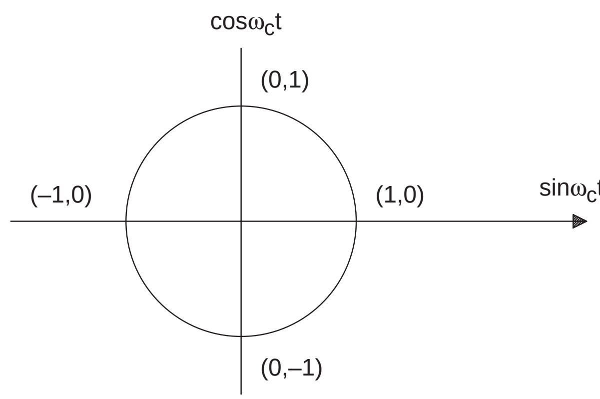


Figure 11 shows that for certain transitions, the origin will have to be crossed. This implies that the power envelope at the decoder will be 0 when the origin is crossed; this can have an undesired impact on the filters. To alleviate this, a $\pi/4$ scheme is used. This is shown in Figure 12. The transitions in this scheme are either +/–45 degrees or +/–135 degrees, and the origin is never traversed in transition from one state to another. This results in eight points on the circle, as shown in Figure 12.

Figure 12. π/4 Differential Quaternary PSK Modulator States

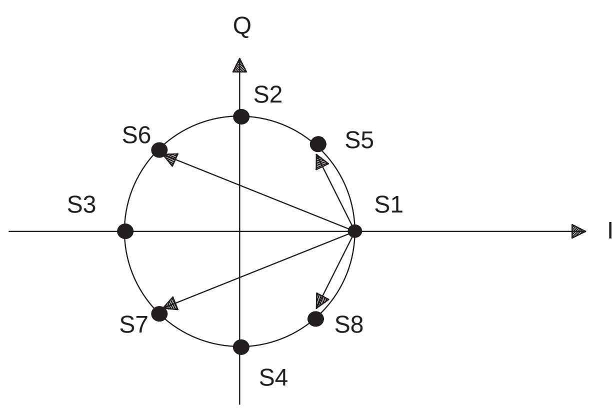
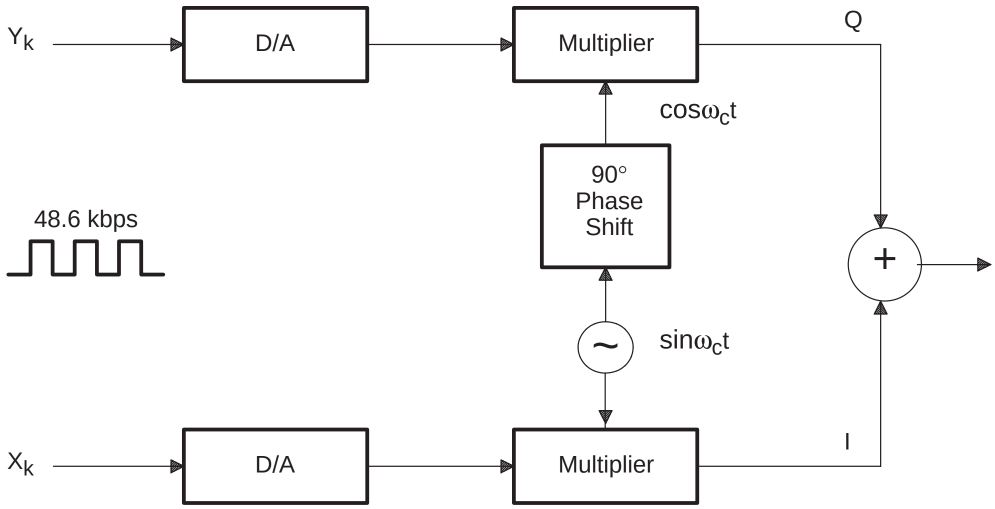


Figure 13 shows how the input serial data is now presented as 2-bit parallel data and is supplied to the multipliers after digital-to-analog conversion. Since two digital-to-analog converters (DACs) are needed, they are sometimes referred to as dual DACs. Binary signals vary the phase-shifted signals via the multipliers. Filters limit the impulse response of the binary signals to ensure that the RF carrier occupies the allocated bandwidth. The two signals are then summed together to form the final phase-shifted carrier. The conversion from baseband to RF (that is, frequency translation of the modulated carrier) is typically carried in several stages in order to reach the 800-MHz range.

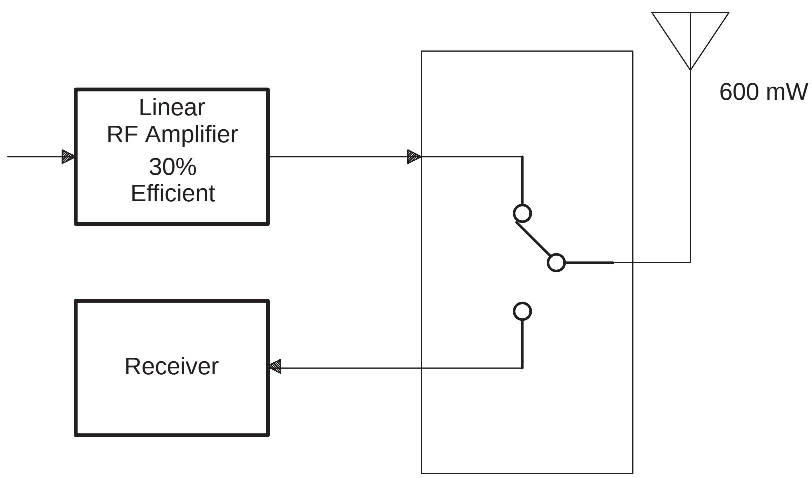
Figure 13. π/4 DQPSK Modulator



RF Amplifier

The RF amplifier boosts the RF-modulated signal to output levels, as specified by the base station. Unlike analog transmission, which uses FM, the RF amplifier for DQPSK carrier must be linear. In FM, class C push-pull nonlinear amplifiers are used for amplification purposes. These nonlinear amplifiers are efficient (about 50%) in order to conserve power. However, nonlinear amplifiers cannot be used in DQPSK, because they would cause phase distortion. Linear amplifiers used for DQPSK are less efficient (30%). Figure 14 shows an RF amplifier.

Figure 14. Linear RF Amplifiers Are Needed for IS-54 Cellular Phone



While a duplexer is required for the analog section of the dual-mode phone, it is not required for the digital portion, because in this case the transmitter and the receiver do not operate simultaneously. A simple PN switch is enough to isolate the receiver from the transmitter, allowing the duplexer to be removed from the digital portion. Removing the duplexer has added benefits: when DQPSK signals are passed through a duplexer, a phase distortion occurs because of group delay; in addition, there is some power loss, which, in turn, requires a higher-rated power amplifier. Hence, removing the duplexer reduces the rating on the power amplifier, which extends the battery life of the mobile unit.

Receiver

The receiver functions in the following order:

1. Amplifies the received radio signal
2. Superheterodynes the RF signal to a lower workable frequency range
3. Demodulates the signal
4. Equalizes or compensates to mitigate the effects of distortions introduced by the radio channel
5. Detects errors
6. Decodes the speech signal
7. Converts it back into analog form and eventually feeds it to a speaker

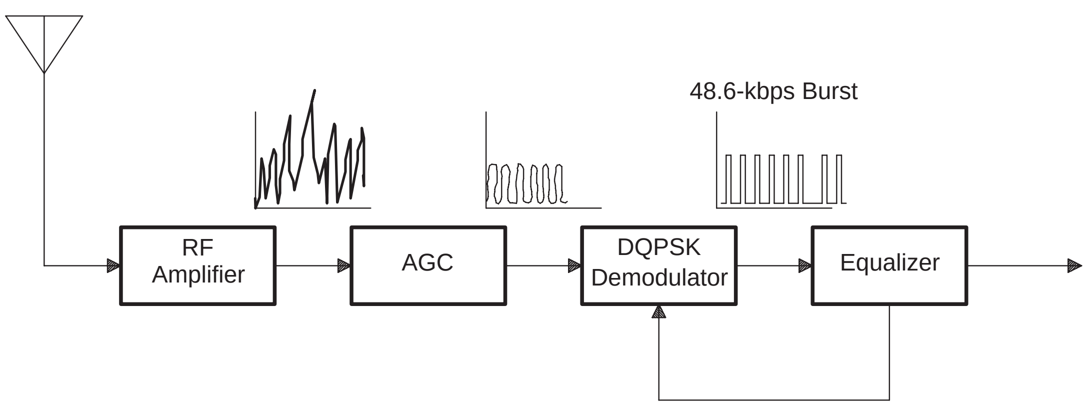
The receiver consists of several functional components:

- Receiver RF amplifier
- Mixer section
- Demodulator
- Channel decoder
- Speech decoder

Receiver RF Amplifier

This section of the receiver amplifies the low-level DQPSK RF carrier, which could be as weak as a few picowatts (~116 dBm). The RF amplifier increases this weak RF signal to a workable range before feeding it to the mixer section. The receiver RF amplifier is a broadband RF amplifier, which has a variable gain controlled by an automatic gain controller (AGC). The AGC compensates for the large dynamic range of the received signal, which is approximately 70 dB. The AGC also reduces the gain of the sensitive RF amplifier so that as the input signal increases, no distortions due to overdriving the receiver occur. Figure 15 shows the RF portion of the receiver.

Figure 15. RF Portion of Receiver Section of Dual-Mode Cellular Phone



Mixer

The frequency of the received carrier is in the range of 869–894 MHz. It is not cost-effective to directly demodulate this RF signal at this frequency range. Typically, the received signal is stepped down to a lower frequency, called the intermediate frequency (IF), by mixing it with a local oscillator (refer to Figure 2). The oscillator source may be varied so that the IF is a constant frequency, which simplifies the IF amplifier design. Typically, a second mixer superheterodynes the first IF with another oscillator source to produce a much lower frequency than the first IF. *A lower frequency enables the design and use of narrow-band filters.*

Demodulator

A DQPSK demodulator extracts data from the IF signal. Typically, a local oscillator with a 90-degree phase-shifted signal is used. The demodulator determines which decision point the phase has moved to; it then determines which symbol is transmitted by calculating the difference between the current phase and the last phase (note that the transmitter is a differential modulator).

Once the symbol has been identified, the next step is to decode the two bits. However, due to noise, Doppler effects, and Rayleigh fading, the signal must be compensated or equalized. Fading occurs when the same RF signal arrives at the receiver at different times because of multiple paths caused by reflections. The Doppler effect is caused by the motion of the transmitter relative to the received signal. The Doppler effect causes the received frequency to vary in proportion to the speed at which the mobile unit is moving; this implies that the equalizer section of a personal communication systems (PCS) unit need not be as complex when it is traveling at pedestrian speeds as when it travels at higher vehicular speeds.

Equalizer

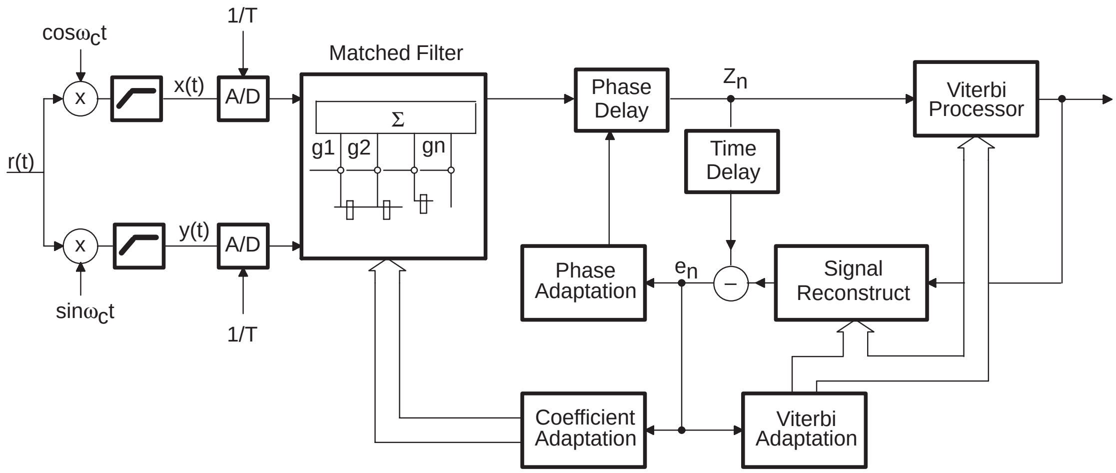
The equalizer is effectively an inverse filter of the channel distortion. Since the RF channel is not constant (as a wireline channel is assumed to be), it is necessary to track or adapt to the changing RF channel. Hence the name *adaptive equalizer.*

The IS-54 specification does not recommend a specific equalizer algorithm. At present, two classes of equalizers are popular:

- The decision feedback equalizer (DFE)
- The maximum likelihood sequence estimator (MLSE)

Figure 16 shows an example MLSE adaptive equalizer [4]. It operates adaptively in a training mode at the beginning of each burst, as well as in a tracking mode during message detection. It includes a matched filter and a modified Viterbi processor. The equalizer in Figure 16 is used by the European GSM system but is similar to the ones used in North America.

Figure 16. An MLSE Adaptive Equalizer

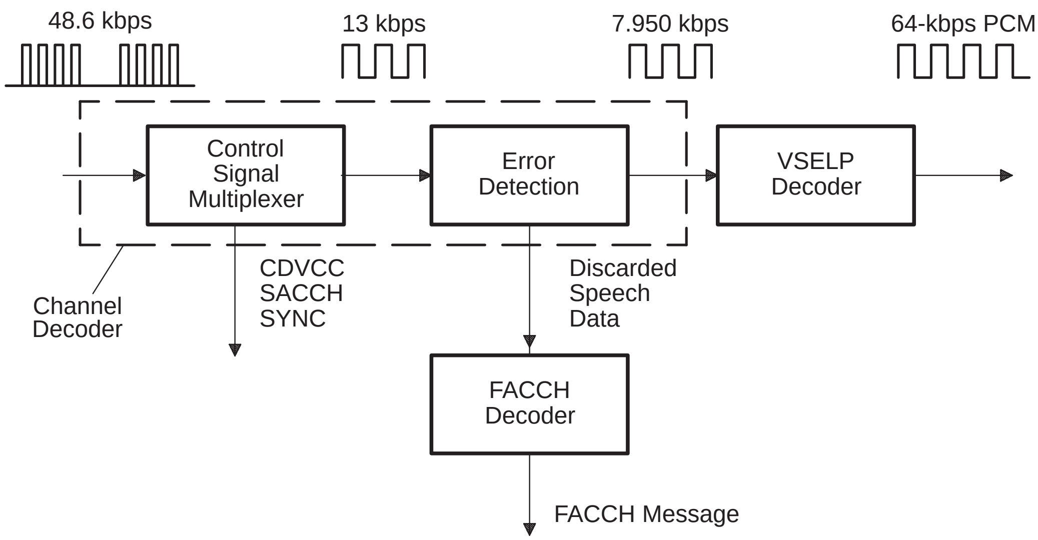


After demodulation and low-pass filtering of the received signal, the components x(t) and y(t) are sampled and A/D is converted, with a sampling frequency equal to the bit rate. Then the signal samples are filtered through a digital N-tap transversal filter, which approximates the matched filter (MF) shown. Theoretically, an MF makes the receiver insensitive to the carrier and clock phases used to demodulate and sample the received signal, provided that the MF coefficients are properly adjusted and the time span of the MF is long enough to include all the channel impulse responses. To this end, you must choose the number of taps, N, in the MF to comply with the maximum number of echo delays that you expect to observe in the operational environment. Note that the modulator output pulses are spread over three bit periods. Typically, N = 6 seems to suffice. The MF output samples are finally processed according to the modified Viterbi processor, which operates on a number of states $S = 2^N - 1$. The complexity of the Viterbi processor varies exponentially with respect to N.

Channel Decoder

The channel decoder detects errors in the bit stream, demultiplexes the control data, and feeds the data to the speech decoder. This is shown in Figure 17. If errors are detected, a masking strategy, explained in *Bad Frame-Masking Strategy* on page 28, is applied.

Figure 17. Channel Decoding and Speech Decoding



The channel decoder works in the following stages:

1. Control signal demultiplexer
2. Error detector

Control Signal Demultiplexer

Speech, SACCH, FACCH, and DVCC data signals from the demodulator are demultiplexed to separate the various signaling information. SACCH and DVCC data are simply demultiplexed by directing the dedicated bits from each burst to their control-processing locations. Speech and FACCH demultiplexing is, however, more challenging. Since FACCH data may replace speech data at any time, FACCH data is extracted by first attempting to detect errors in speech data. If the CRC appears to be correct as decoded for a speech slot, the data is routed to the speech codec section. When the CRC is in error, the data is then decoded as a FACCH message. If the CRC appears to be correct, this FACCH message is routed to its call-processing location.

Error Detector

DVCC words are error-detected, compared to the assigned DVCC to determine cochannel interference, and sent to the transmit section to be echoed back to the base station.

The channel decoder provides BER information and RSSI when commanded by the base station. This feature is called MAHO, which is discussed in the *Mobile Assisted Handoff* section on page 9.

Bad Frame-Masking Strategy

The bad frame-masking strategy is based on a 6-state machine. On every decode of a speech frame, the state machine can change states. State 0 occurs most often and implies that the CRC comparison was successful. State 6 implies that there were at least six consecutive frames that failed the CRC check. The action taken at each of these states varies as well. At state 0, no action is taken. States 1 and 2 are simple frame repeats. States 3, 4, and 5 repeat and attenuate the speech. State 6 completely mutes the speech. A detailed description of the action corresponding to each state follows:

- *State 0*: No CRC error is detected. The received decoded speech data is used.
- *State 1*: A CRC error detected. Parameter values R(0) and the LPC bits from the last frame that was in state 0 are repeated. The remaining decoded bits for the frame are passed to the speech decoder without modification.
- *State 2*: Identical to the action for state 1.
- *State 3*: Similar to the action for state 1, except that the value for R(0) is modified. A 4-dB attenuation is applied to the R(0) parameter: that is, if R(0) of the last state 0 frame is greater than 2, then R(0) is decremented by 2 and repeated at this lower level.
- *State 4*: Similar to state 3. A further attenuation by 4 dB is applied to R(0) so that the level is as much as 8 dB from the original value of R(0).
- *State 5*: Similar to 4. R(0) is further attenuated by 4 dB.
- *State 6*: The frame is repeated; but this time R(0) is cleared to 0, totally muting the output speech. Alternatively, comfort noise could be inserted in place of the speech signal.

Speech Decoder

The speech decoder, VSELP, converts the 7950-bps input data stream into 64-kbps PCM data. In poor radio conditions, the performance of VSELP has been shown to be superior to analog cellular. This is primarily due to the error-protection and error-detection capabilities that are made possible by digital techniques.

When speech frames are lost because of errors and are not correctable, the speech coder repeats the previous frame information. If the number of consecutive lost speech frames increases, a gradual muting is applied. Thus, gaps are filled by using the characteristics of the human ear.

When the user data is not speech, but computer or facsimile data, then the speech decoder is bypassed.

Adaptive Spectral Postfilter

The perceptual quality of the synthetic speech can be enhanced by using an adaptive spectral postfilter as the final processing step. The form of the postfilter is

$$H(z) = \frac{1 - \sum_{i=1}^{10} n_i z^{-i}}{1 - \sum_{i=1}^{10} v^i \alpha_i z^{-i}} \quad 0 \le v < 1$$

α_i = *Coefficient of synthesis filter*

Audio Interface

The output of the speech coder, a 64-kbps bit stream, is input to the audio interface, which consists of the following stages:

1. Digital-to-analog conversion
2. Reconstruction filter
3. Receive-level adjustment

The reconstruction filter minimizes the step transients caused by the D/A converter. The receive-level sensitivity is defined so that a value of 24 in the R0 field, the frame energy, causes an acoustic level of at least 97 dB at the transducer when measured by an artificial ear. R0 equal to 24 represents the average frame energy during a frame, which is 18 dB down from full scale.

Summary

This report presents a brief functional overview of a digital cellular mobile station. Emphasis is given to the algorithmic description and implementation aspects of each function. The main purpose of this paper is to provide a general introduction to various functional blocks. Refer to the other papers in this book for a detailed implementation description of the individual functions.

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