Comparisons Between USDC (IS-54) and GSM

Surachet Kanprachar

Department of Electrical and Computer Engineering, Faculty of Engineering,
Naresuan University, Phitsanulok 65000, Thailand.
Corresponding author. E-mail address: skanprac@vt.edu (S. Kanprachar)
Received 29 April 2004; accepted 8 November 2004

Summary

United States Digital Cellular (USDC) system and Global System for Mobile Communication (GSM) are two main standards of the second generation of digital wireless telephone system. Both of them have similarities and differences. In this paper many important characteristics of these two standards are discussed. The details of USDC and GSM are investigated. Focusing on the modulation types, the effect of pulse shapings, Bit-Error-Rate (BER), and spectral efficiency will be compared by using computer graphic analysis. USDC has better performances than GSM in terms of zero intersymbol interference (ISI) condition and spectral efficiency. However, there is one advantage of GSM over USDC; that is, the power efficiency. Both of them have the same characteristic of BER over a Gaussian channel.

Introduction

The second generation of wireless cellular telephone was issued to increase the capacity, improve performance, and add some new services to the previous analog system. To achieve these goals, the digital techniques are employed. There are many standards supporting this requirement. Two of them are United States Digital Cellular (USDC) system Interim Standard-54 (IS-54), which was issued for North America, and Global System for Mobile Communication (GSM), which was issued for Europe. Comparing these two standards, there are some similarities and differences.

USDC and GSM use the same multiple access technique; that is, Time Division Multiple Access (TDMA/FDD) / Frequency Division Duplex (FDD) while there are many differences between them such as the modulation type, frame structure, speech coding etc. In this paper, some characteristics of these two standards will be studied intensively. In the following sections, the details of USDC and GSM will be investigated. These two standards use different types of modulation. Because of these different modulation types, some characteristics of these two standards are different. The effect of pulse shapings, Bit-Error-Rate (BER), and spectral efficiency will be compared. The comparisons are based on the computer graphic analysis using the information from the previous sections.

The US Digital Cellular System (USDC)

The US digital cellular system (USDC) is also known as Digital Advance Mobile Phone Service (D-AMPS) because it was designed to use the same channel allocations, frequency reuse plan, and base stations as

Advance Mobile Phone Service (AMPS), which is the first generation analog system. However, USDC, which is the second generation of the cellular system in the US, needs to increase capacity, improve the performance, and add, if possible, new innovative services (in addition to the telephone) (Feher, 1991), so the digital technology is adopted to support these requirements. The entire transmission standard for USDC is Interim Standard 54 (IS-54), which was accepted by the Electronic Industry Association and Telecommunication Industry Association (EIA/TIA) on 1990 (Goodman, 1991). IS-54 is the dual mode system standard that allows both AMPS and USDC services to be offered in the same network; that is, the dual mode system. The general description for USDC is focused on this section. Also, the digital modulation type of USDC is studied.

Frequency allocation and channel bandwidth

As mentioned previously, the USDC uses the same channel allocations as AMPS; therefore, the reverse channel frequency band for USDC is from 824 to 849 MHz, and the forward channel frequency band for USDC is from 869 to 894 MHz. The frequency spacing between the reverse and forward channels is 45 MHz. Each channel covers the bandwidth of 30 kHz. There are 832 channels in total. The control channels for USDC are analog control channels, as used in AMPS (Raith and Uddenfeldt, 1991).

Multiple access

Duplexing in USDC is Frequency Division Duplexing (FDD), which provides two distinct bands of frequencies for every user. In USDC, these two distinct bands are 45 MHz apart, as described above. Time Division Multiple Access (TDMA) is the multiple access method for USDC. USDC divides the channel bandwidth into time slots. Each time slot is allowed for the user to transmit or receive the signal. Since each channel bandwidth is divided into time slots, many users can be supported by one channel bandwidth; thus, the capacity of the system is increased. Each channel bandwidth of USDC can support up to 6 users with half-rate speech coder of 3.975 kbps/user (Rappaport, 1996).

Frame structure

In USDC, six time slots are combined to be one TDMA frame of 40 ms frame duration, as shown in Figure 1. Each time slot covers 6.67 ms of time duration. One frame of USDC contains 1944 bits (Goodman, 1991), thus, the data rate is 48.6 kbps. One time slot carries 324 bits including 260 bits of user information and 12 bits of system control information or slow associated control channel (SACCH). The remaining 52 bits carry 28 bits of time synchronization (SYNC), 12 bits of digital verification color code (DVCC). In the mobile-to-base station time slot, the remaining 12 bits are a 6-bit guard time interval, when no energy is transmitted, followed by a 6-bit ramp interval to allow the transmitter to reach its full output power level. In the base station-to-mobile time slot, these 12 remaining bits are reserved for future use (RSVD).

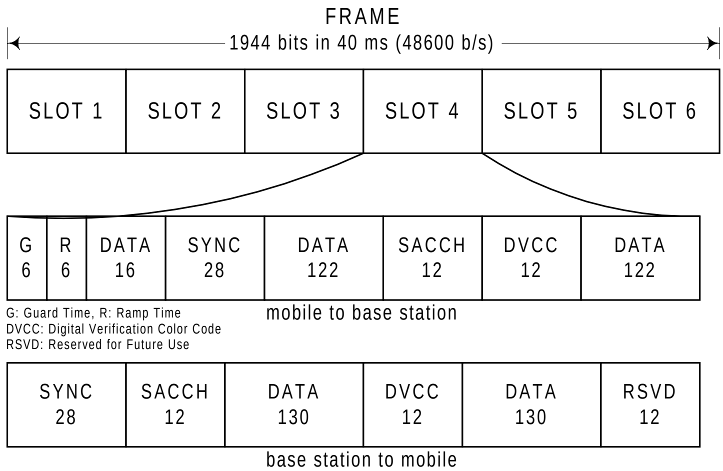

Figure 1 IS-54 slot and frame structure (Goodman, 1991)

The user information bits are divided into two 122-bit data block and one 16-bit data block, in mobile-to-base station direction. In base station-to-mobile direction, the information bits are divided into two 130-bit data block. The slow associated control channel (SACCH) is sent in every time slot, providing a signaling channel parallel with the digital speech. The mobile assisted handoff (MAHO) is also supported by SACCH; that is, SACCH from mobile unit reports the results of signal strength measurements of neighboring base stations. When the received power from the base station of a neighboring base station begins to exceed the received power from the current base station by a certain value or for a certain period of time, the handoff is initiated (Rappaport, 1996).

The 28-bit of synchronization signal contains a known bit pattern that allows the receiver to establish bit synchronism and to train an adaptive equalizer (Goodman, 1991). Six different synchronization patterns are specified by the system, one for each time slot in the TDMA frame. The digital verification color code (DVCC) is a 12-bit message, which is sent in every time slot. There are 256 color codes generated by 8 bits. These color codes are protected by (12,8) Hamming code; thus, DVCC contains 12 bits. One of these codes is assigned to each base station to prevent a receiver from locking onto an interfering signal from a distance cell.

Speech coding

The speech coding adopted for USDC is a vector-sum excited linear predictive coder (VSELP). This speech coder belongs to the class of code excited linear predictive coder (CELP) (Rappaport, 1996). The block diagram of the VSELP speech coder is shown in Figure 2. The coder derives a new linear predictor every 20 ms. This linear predictor is characterized by ten log-area ratios, which are represented by 38 bits. The bits for frame energy are 5 bits. For long-term predictor, it generates coefficients at 5-ms intervals, four times per 20-ms block. The log of each long-term predictor is represented by 7 bits, accounting for 28 bits per 20 ms and 8 bits of gain information; thus, total bits for 20-ms block are 88 bits. All of these bits are summed up to 159 bits per 20-ms block or 7950 bps.

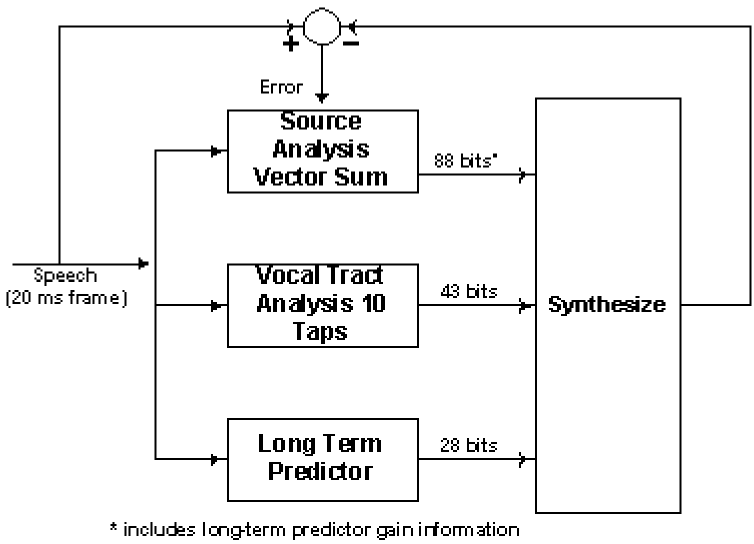

Figure 2 Vector sum excited linear prediction speech coder (Goodman, 1991)

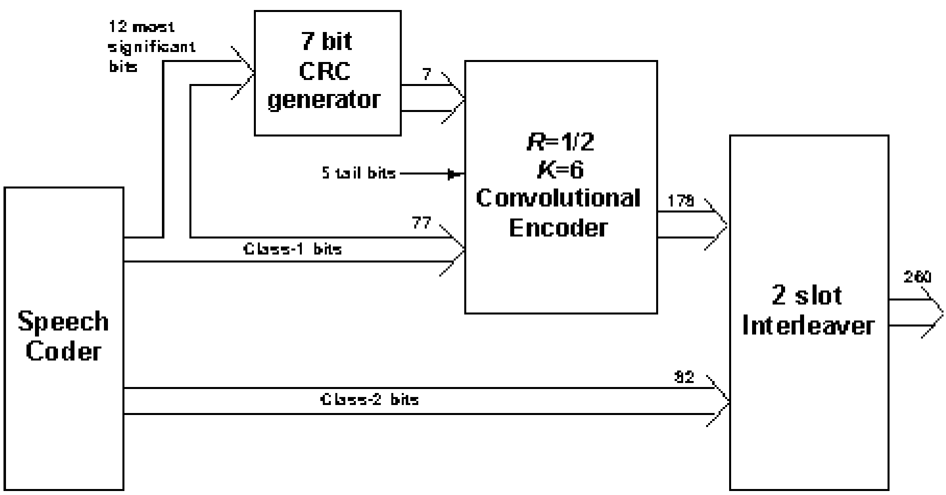

Figure 3 Error protection for USDC speech coder output (Rappaport, 1996)

Channel coding

The output bits of the VSELP speech coder are divided into two groups; one is 77 class-1 bits and the other is 82 class-2 bits. The 77 class-1 bits have a greater influence on speech quality than the 82 class-2 bits; therefore, these 77 class-1 bits are protected by an error detecting code and a rate ½ convolutional code to produce a sequence of 178 bits, as shown in Figure 3. The twelve most significant bits of class-1 bit are block coded using a 7-bit CRC error detection code (Rappaport, 1996) to generate output of 7 bits. These 7 bits output from CRC generator, 5 tail bits, and 77 class-1 bits are sent to a rate ½ convolutional encoder using constraint length K of 6. The 178 bits output of the convolutional encoder and 82 class-2 bits are multiplexed together producing 260 transmitted bits. Two blocks of 260 channel bits are interleaved and placed into the two assigned time slots of the 40-ms transmission frame (Goodman, 1991).

Modulation technique

In USDC, π/4 DQPSK modulation technique has been adopted. This type of modulation was first proposed for data transmission via telephone lines by Baker of Bell Telephone Laboratories (Feher, 1991). A π/4 DQPSK is used with 0.35 rolloff raised cosine filtering to transmit the gross data of 4.6 kbps within 30 kHz channel bandwidth. Since it is differentially encoded at the transmitter, the detector of this modulation type can be coherent, differential, or discrimination detector. Certainly, on the receiver side, there must be a raised cosine filter that matches that of the transmitter side for satisfaction of zero intersymbol interference (ISI) condition.

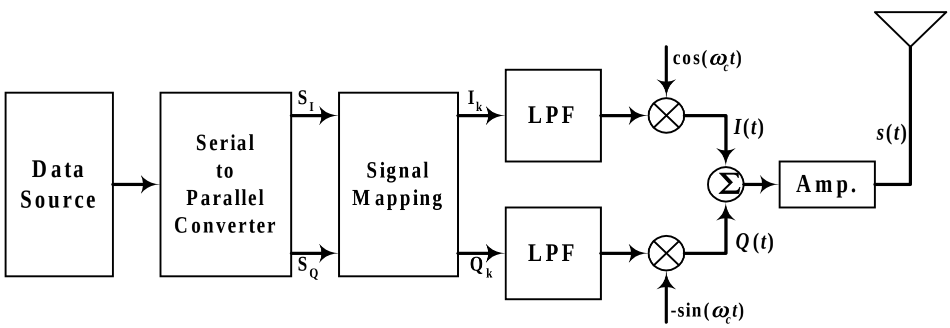

Figure 4 Block diagram of the transmitter of π/4 QPSK modulator (Yacoub, 1993)

The block diagram of π/4 QPSK is shown in Figure 4. Note that the transmitters of π/4 DQPSK and π/4 QPSK are identical except that the input data bits for π/4 DQPSK are differential encoded. Thus, in this section, the operation of π/4 QPSK is considered. The binary data are first converted to two parallel data streams referring to S_I and S_Q. Then, these two data are mapped by the signal mapping circuit to be the in-phase (I_k) and quadrature (Q_k) pulse amplitudes. These pulse amplitudes are determined from their previous values (I_{k-1} and Q_{k-1}) and an absolute phase angle (θ_k) (Rappaport, 1996), as shown in equation (1) and (2). The absolute phase angle (θ_k) for the symbol k^{th} is a function of its previous value (θ_{k-1}) and a phase shift (ϕ_k), which is a function of the current input data (S_I and S_Q).

$$I_k = \cos\theta_k = I_{k-1}\cos\phi_k - Q_{k-1}\sin\phi_k \quad (1)$$

$$Q_k = \sin\theta_k = I_{k-1}\sin\phi_k + Q_{k-1}\cos\phi_k \quad (2)$$

$$\theta_k = \theta_{k-1} + \phi_k \quad (3)$$

The relationship between ϕ_k and the current input data is given in Table 1. The phase shift (ϕ_k) can only be $k\pi/4$ where k is ±1 or ±3. From equation (1), (2), and (3), it is seen that I_k and Q_k can take the amplitudes of ±1, 0, and $\pm 1/\sqrt{2}$. The signal constellation of a π/4 QPSK is shown in Figure 5.

Table 1 Carrier Phase Shifts Corresponding to Various Input Bit Pairs

Information bits S_I,S_Q	Phase shift ϕ_k
1 1	π/4
0 1	3π/4
0 0	-3π/4
1 0	-π/4

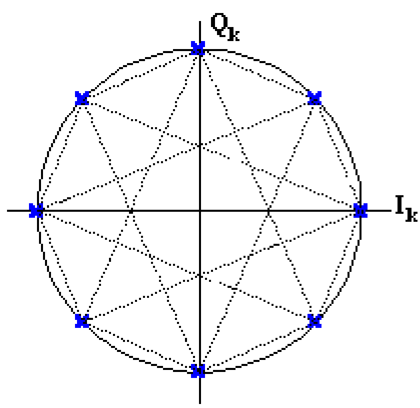

Figure 5 Constellation diagram of a p/4 QPSK signal

The signal I_k and Q_k are sent through lowpass filters, and, then, they will be modulated to the carrier frequency (f_c) to be $I(t)$ and $Q(t)$. The in-phase ($I(t)$) and quadrature ($Q(t)$) signals are summed together and amplified to be the transmitted signal, $s(t)$. The transmitted signal can be given by

$$s(t) = I(t)\cos\omega_C t - Q(t)\sin\omega_C t \tag{4}$$

It is seen that the information in a π/4 QPSK signal is contained in the phase difference ϕ_k of the carrier between two adjacent symbols. At the receiver the phase difference between two sampling instants is needed only; thus, the differential detection can be employed.

In USDC, the raised cosine pulse shaping is adopted with a rolloff factor of 0.35. This type of pulse shaping satisfies the zero ISI condition. The raised cosine rolloff transfer function can be achieved by using identical $\sqrt{H_{RC}(f)}$ filters at the transmitter and receiver sides. The expression for this filter is shown in equation (5) and (6) below (Rappaport, 1996).

Frequency domain:

$$H_{RC}(f) = \begin{cases} 1 & ;0 \le |f| \le (1-\alpha)/2T_S \\ \frac{1}{2}\left[1+\cos\left(\frac{\pi[(2T_S|f|)-1+\alpha]}{2\alpha}\right)\right] & ;(1-\alpha)/2T_S < |f| \le (1+\alpha)/2T_S \\ 0 & ;|f| > (1+\alpha)/2T_S \end{cases} \tag{5}$$

where α is the rolloff factor which ranges between 0 and 1.
T_s is the symbol period.

Time domain:

$$h_{RC}(t) = \left(\frac{\sin(\pi t/T_S)}{\pi t}\right)\left(\frac{\cos(\pi\alpha t/T_S)}{1-(4\alpha t/(2T_S))^2}\right) \tag{6}$$

The power spectral density of Quadrature Phase Shift Key (QPSK) signal depends on the power spectral density of the pulse shaping function. For QPSK, there are four possible signals to be transmitted. The probabilities of sending these signals are identical. Thus, the power spectral density of QPSK signal is the same profile as the power spectral density of the pulse shaping function except the center frequencies are shifted to be at $\pm f_c$. The differential encoding does not modify the power spectral density, so the power spectral density of a Differential Quadrature Phase Shift Key (DQPSK) signal is the same as that of a QPSK signal (Couch, 2001).

Since π/4 DQPSK can be detected by using coherent detection or differential detection, the bit error probability calculation of it is divided into two cases. One is for coherent detection, the bit error calculation is shown in equation (7) (Couch, 2001). For a differential detection, the bit error probability calculation is also shown in equation (9) (Couch, 2001). These two equations are for the signal with Gray coding over a Gaussian channel.

Coherent detection:

$$P_b = Q\left\{\frac{2E_b}{N_0}\right\} \tag{7}$$

where

$$Q\{x\} = \int_x^\infty \frac{1}{\sqrt{2\pi}} e^{-\frac{y^2}{2}} dy = \frac{1}{2} erfc\left(\frac{x}{\sqrt{2}}\right) \tag{8}$$

Differential detection:

$$P_b = \int_b^\infty x \exp\left(-\frac{a^2 + x^2}{2}\right) I_0(ab)dx - \frac{1}{2}\exp\left(-\frac{1}{2}(a^2 + b^2)\right) I_0(ab) \tag{9}$$

where $I_0(\cdot)$ is the modified Bessel function of order zero and

$$a = \sqrt{\frac{E_b}{2N_0}}\left(\sqrt{2+\sqrt{2}} - \sqrt{2-\sqrt{2}}\right) \tag{10}$$

$$b = \sqrt{\frac{E_b}{2N_0}}\left(\sqrt{2+\sqrt{2}} + \sqrt{2-\sqrt{2}}\right) \tag{11}$$

E_b is the energy contained in one bit period
N_0 is the noise power

For the fading channel, the bit error probability calculation of the QPSK is given by equation (12). This equation is for the slow fading channel. Equation (13) is the expression for finding the bit error probability of a DQPSK signal in the fast fading channel (Couch, 2001). These two equations will be plotted and discussed in the last section.

QPSK (slow fading):

$$P_b = \frac{2}{3}\left\{1-\left(1-\frac{1}{2}(1-\sqrt{\frac{E_b/N_0}{1+E_b/N_0}})\right)^2\right\} \tag{12}$$

DQPSK (fast fading):

$$P_b = \frac{2}{3}\left\{1-\left(1-\frac{1}{2}(1-F)\right)^2\right\} \tag{13}$$

where

$$F = \frac{J_0(2\pi f_D T)}{\sqrt{2(1+(E_b/N_0)^{-1})^2 - J_0{}^2(2\pi f_D T)}} \tag{14}$$

$J_0(\cdot)$ is the zeroth-order Besssel function of the first kind
$J_0(2\pi f_D T)$ is the normalized envelope correlation
f_D is the maximum Doppler frequency.

The Global System of Mobile Communication (GSM)

Frequency allocation and channel bandwidth

The Global System of Mobile communications (GSM) is a digital cellular communication system initially developed in a European. For GSM-900 system, two frequency bands have been made available; that is, 890 - 915 MHz for uplink (direction to Mobile Station (MS) to Base Station (BS) and 935 - 960 MHz for downlink (direction BS to MS). By using Frequency Division Multiple Access (FDMA), the bandwidth of 25 MHz is divided into 124 pairs of frequency duplex channels with 200 kHz carrier spacing and duplex spacing of 45 MHz between uplink and downlink directions. A guard band of 200 kHz is left between the bottom edge of each band and the first RF carrier. The carrier frequencies in the two bands for n^{th} duplex radio channel will be 890.2 + 0.2(n - 1) MHz for uplink and 935.2 + 0.2(n - 1) MHz for downlink.

Multiple access

Time Division Multiple Access is used to split a 200 kHz radio channel into 8 time slots. In other word, it creates 8 logical channels. Each channel transmits the digitized speech in a series of short bursts, so a GSM terminal is only transmitting for one eighth of the time. The 8-slot TDMA together with 124 physical full duplex channels corresponds to total 992 logical full duplex channels.

Frame structure

The GSM system distinguishes between traffic channels and control channels. Traffic Channel/Full-Rate Speech (TCH/FS) is used to carry speech at the speed of 13 kbps. In the TCH, data are transmitted in bursts (148 bits), which are placed in time slots. The 8.25 bit guard time allows for some propagation time delay in the arrival of bursts. A total of 156.25 bits is transmitted in 0.577 milliseconds (bit period 3.79 microseconds), giving a gross bit rate of 270.833 kbps. Eight bursts of eight users are multiplexed onto one RF carrier giving a TDMA frame of 8 x 0.577 ms ≈ 4.615 ms. The effective information throughput is 114/4.615 ms ≈ 24.7 kbps which is sufficiently high to transmit TCH/FS.

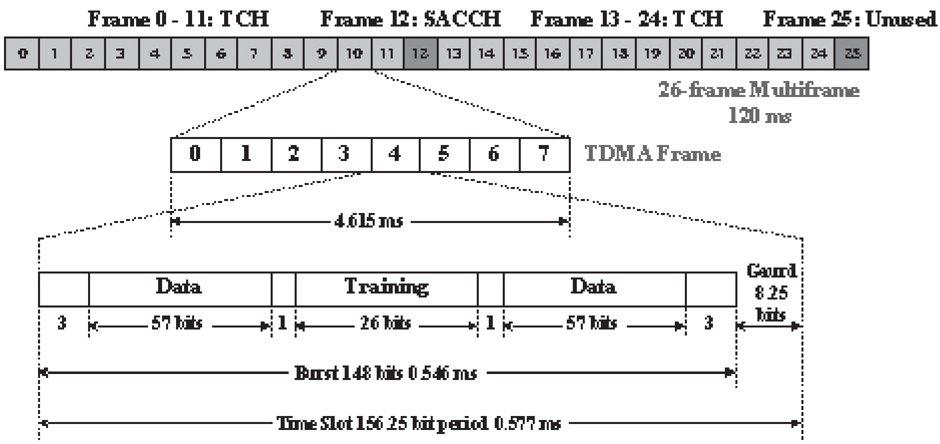

Figure 6 The TDMA frame structure of GSM system (Raith and Uddenfeldt, 1991)

The TDMA frame of eight users are multiplexed together into multiframes of 24 TDMA frames, but the 13th will carry a Slow Associated Control Channel (SACCH) message, while the 26th will be an idle or dummy frame. The 24 TCH/FS frames are sent in a multiframe during 26 x 4.615 ms ≈ 120 ms. The traffic throughput is reduced to 24/26 x 24.7 kbps ≈ 22.8 kbps.

Speech coding

The full rate speech codec in GSM is described as Regular Pulse Excitation with Long Term Prediction (GSM 06.10 RPE-LTP). The encoder divides the speech into short-term predictable parts, long predictable parts and the remaining residual pulse. Then, it encodes that pulse and parameters for the two predictors. Information from previous samples, which does not change very quickly, is used to predict the current sample. The coefficients of the linear combinations of the previous samples, plus the encoded form of the residual, the difference between the predicted and actual sample represent the signal. Speech codec produces 260 bits for every 20 milliseconds, giving a total bit rate 13 kbps.

Channel encoding

The 260 bits of the speech block are classified into two groups. The 78 Type II bits are considered of less importance and are unprotected. The 182 Type I bits are split into 50 Type Ia bits and 132 Type Ib bits. Type Ia bits are first protected by 3 parity for error detection. Type Ib bits are then added together with 4 tail bits before applying the convolutional code with rate r = 0.5 and constraint length K = 5. The GSM convolutional code consists in adding 4 bits to the initial 185-bit sequence. It applies two different convolutions. The result is composed of twice of 189-bit sequence. The GSM convolutional coding rate per data flow is 378 bits / 20 ms ≈ 18.9 kbps. The resulting 378 bits are then added to the Type II bits. It produces a complete coded speech frame of 456 bits. The GSM bit rate flow is 456 bits / 20 ms ≈ 22.8 kbps.

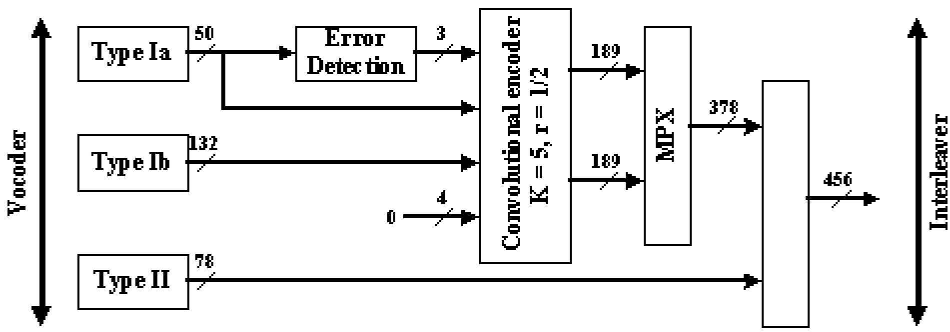

Figure 7 GSM's channel coding

Full rate speech blocks are interleaved on eight bursts. The 456 bits are divided into 8 blocks of the sub-blocks of 57 bits each. A sub-block is defined as odd and even numbered bits. Each of them is carried by a different burst and in a different TDMA frame. Since each time-slot burst can carry two 57-bit blocks, each burst carries traffic from two different speech samples. So, a burst contains the contribution of the two successive speech blocks.

Modulation technique

Each time slot burst is transmitted at a bit rate 270.833 kbps. This digital signal is modulated onto the analog carrier frequency, which has bandwidth of 200 kHz. Gaussian Minimum Shift Keying (GMSK) is used with modulation index of 0.5 and *BT* (bandwidth times bit period) of 0.3. GMSK was selected by GSM over the other modulation schemes as a compromise between fairly high spectrum efficiency and reasonable demodulation complexity. Compared to MSK, Gaussian pulse shaping filter can considerably reduce the side-lobe level in the transmitted spectrum. Furthermore, GMSK is efficient power because of the phase modulation and its constant envelope. The constant envelope allows the use of simple power amplifiers and the low out-of-band radiation minimizes the effect ofadjacent channel interference. GMSK modulator can be seen in Figure 8.

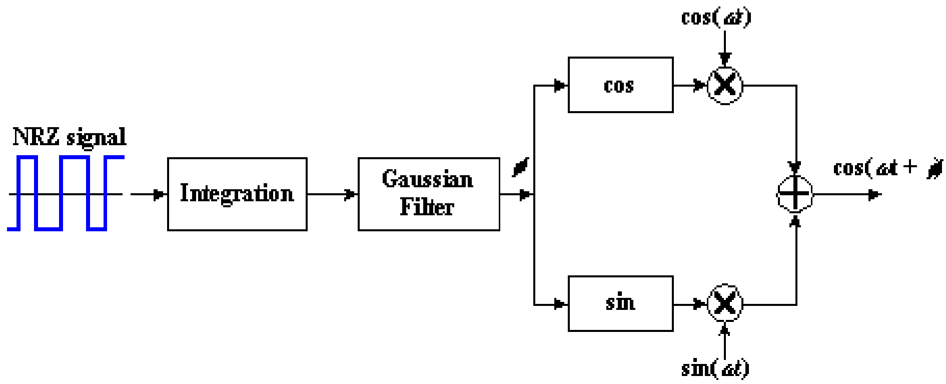

Figure 8 GMSK modulator scheme (Steele, 1992)

GMSK modulation can be implemented by using a Gaussian pre-modulation low pass filter and FM modulator. GMSK signal is produced by applying the NRZ data stream of unity amplitude to the Gaussian filter. The time-domain impulse response of the filter and its transfer function are given in equations (15) and (16), respectively.

$$h_G(t) = \frac{\sqrt{\pi}}{\alpha} \exp\left[-\frac{\pi^2}{\alpha^2} t^2\right] \qquad (15)$$

$$H_G(f) = \exp\left[-\alpha^2 f^2\right] \qquad (16)$$

where

$$\alpha = \frac{\sqrt{\ln 2}}{B\sqrt{2}} \qquad (17)$$

B is the 3-dB bandwidth of a low pass filter having a Gaussian shaped spectrum
T is the bit period, and
$B_N = BT$ is the normalized bandwidth.

By convolving a NRZ data stream of the unity amplitude with this filter, it is found that the output is given by (Steele, 1992)

$$g(t) = \frac{1}{2T}\left[Q\left(2\pi B \frac{t - T/2}{\sqrt{\ln 2}}\right) - Q\left(2\pi B \frac{t + T/2}{\sqrt{\ln 2}}\right)\right] \qquad (18)$$

The measured static BER performance in the non-fading environment can be approximated as (Rappaport, 1996)

$$P_b = Q\left\{\sqrt{\frac{2\delta E_b}{N_0}}\right\} \tag{22}$$

where

$$\delta = \begin{cases} 0.68 \text{ ; for GMSK with } BT = 0.25 \\ 0.85 \text{ ; for simple MSK } (BT = \infty) \end{cases}$$

In the slow fading, BER is given by (Rappaport, 1996)

$$P_b = \frac{1}{2}\left(1 - \sqrt{\frac{\delta\Gamma}{\delta\Gamma + 1}}\right) \cong \frac{1}{4\delta\Gamma} \tag{23}$$

where Γ is the average signal-to-noise ratio.

Comparisons between USDC (IS-54) and GSM

In this section, pulse shaping, BER, and spectral efficiency of USDC and GSM are compared. The details of USDC and GSM explained in the previous sections are used in this section. Advantages and disadvantages of techniques used in these two systems are discussed.

Pulse Shapings

USDC and GSM use different pulse shapings, so their power spectral densities are different. USDC uses a raised cosine rolloff pulse shaping with a rolloff factor of 0.35 while GSM uses a Gaussian pulse shaping with a bandwidth-bit period (*BT*) product of 0.3. With the raised cosine rolloff pulse shaping, USDC gives a much better ISI than GSM because its pulse shaping satisfies the zero ISI condition, as shown below in Figure 9. From this figure, if the received signal is sampled every $k/(2f_0)$ where k is an integer and f_0 is the 3-dB bandwidth of the raised cosine filter, the neighboring symbols do not affect the sampling symbol; thus, zero ISI. For GSM, the Gaussian impulse response for different values of *BT* is shown in Figure 10. If *BT* is small, the sidelobe of the frequency response of the Gaussian pulse is relatively small but the ISI problem then becomes more severe. However, if *BT* is large, the sidelobes of the frequency response of the Gaussian pulse is large; thus, high-frequency cutoff in any devices can cause problem. The advantage of having large *BT* is that the ISI problem is lessened. In GSM system, a BT of 0.3 (shown by the solid line) is adopted. This *BT* does not completely satisfy the zero ISI condition but is still acceptable.

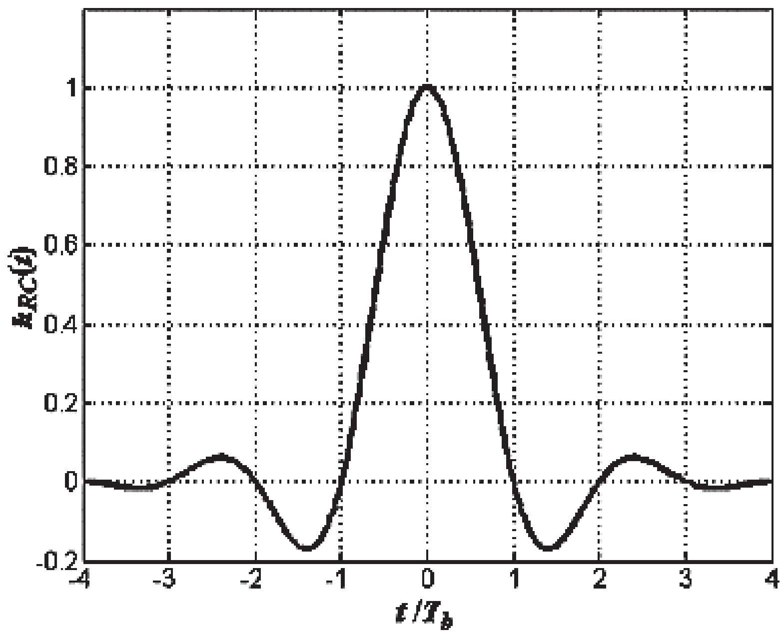

Figure 9 Normalized Impulse Response of a raised Cosine Rolloff Filter with a = 0.35

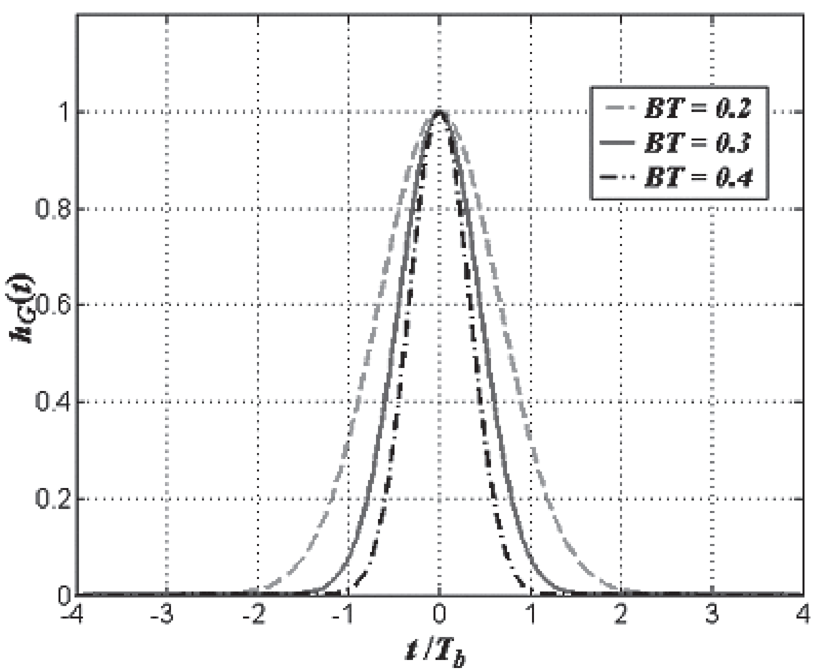

Figure 10 Normalized Impulse Response of a Gaussian Pulse Shape with BT = 0.2, 0.3, and 0.4

Comparing these two pulse shapings, even though the raised cosine rolloff filter gives a better ISI, it requires a linear amplification, which gives low power efficiency. On the other hand, the Gaussian pulse shape has a worse ISI condition but a non-linear amplifier can be used efficiently. The issue of the amplifier efficiency is important especially in the mobile unit. A non-linear amplifier (type C) has 70 percents power efficiency while type A and B amplifiers, which are linear, only have 30-40 percent power efficiency. So by using type C amplifier, the GSM mobile unit can have a better battery lifetime.

Bit Error Rate (BER) or Bit Error Probability (P_b)

For USDC, the coherent and non-coherent detections can be used. Using a coherent detection with a received signal over an Additive White Gaussian Noise (AWGN) channel, the bit error probability can be found using equation (7). With a differential detection, the bit error probability can be found using equation (9). For GSM, both of detection techniques can be used. The bit error probability can be found by using equation (22) with *BT* of 0.25. The Plotting of their bit error probabilities is also shown in the Figure 11.

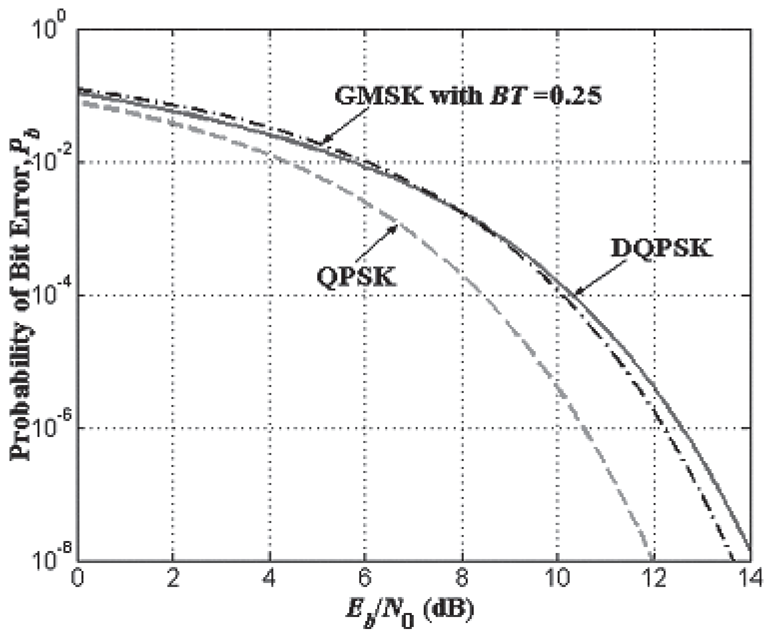

Figure 11 Bit error rate performance of QPSK, DQPSK, and GMSK over an AWGN channel.

From the figure above, the coherent detection for QPSK gives a better BER than others. The BERs for DQPSK and GMSK are approximately the same. At the BER of 10^{-6}, both of them have 2 dB of E_b/N_0 worse than QPSK.

However, the implementation of a coherent detection is more complex than non-coherent detection.

For the fading channel, it is divided into two main types; i.e., fast fading (selective fading) and slow fading (non-selective fading). In the slow fading, the probability of bit error is plotted as a function of the signal to noise ratio (or E_b/N_0) by using equation (12) and (23) for QPSK and GMSK, respectively, as shown in Figure 12. From this figure, it is seen that QPSK and GMSK have almost the same characteristic of bit error rate. They have a linear relation between the E_b/N_0 and Pb. Compared to the fast fading case (i.e., fading DQPSK curves), both of them have a 6 dB better in terms of E_b/N_0.

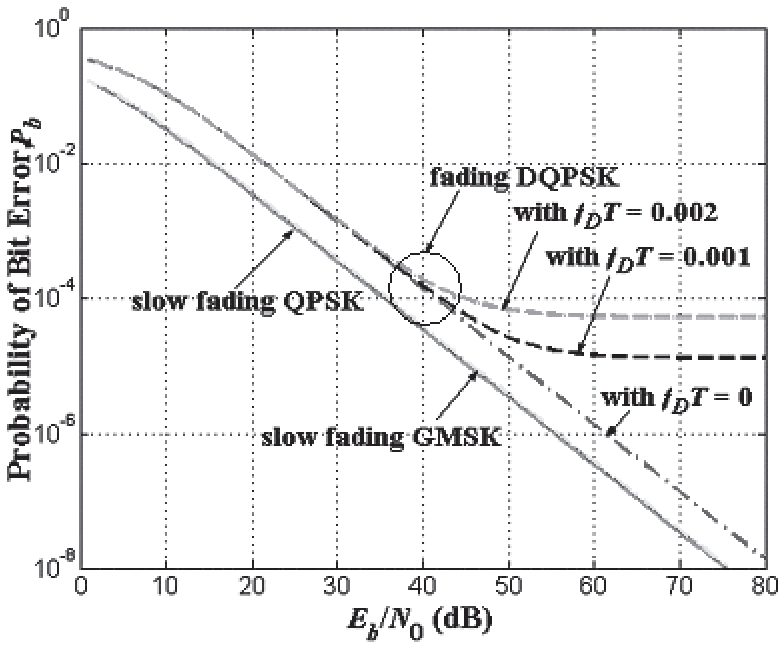

Figure 12 Bit error rate performance of QPSK, DQPSK, and GMSK over a fading channel

In the fast fading, the probability of bit error is also plotted as a function of E_b/N_0 of DQPSK signal by using equation (13). This is also shown in Figure 12. There are three different BER curves corresponding to the different Doppler shift frequencies (thus, different f_DT). These three curves are almost identical for E_b/N_0 lower than 35 dB. If E_b/N_0 is higher than 35 dB, the differences between these three curves are noticeable. For higher Doppler frequency, the BER is degraded significantly as can be seen by the BER floor for the case of f_DT = 0.001 and 0.002. This BER floor means that increasing the signal power will not help improving the BER. For GMSK, the BER in fast fading has not yet been theoretically estimated because the tracking performance of the carrier recovery circuit in such environment cannot be analyzed (Murota, 1981). From Figure 12, it is seen that the probability of bit error increases if the Doppler frequency increases. GMSK in the fast fading has the same condition as the DQPSK. That is, BER floor for high E_b/N_0 will also happen.

Considering the BERs from AWGN and fading channels, it is seen that to get a BER of 10^{-6} for the case of fading channel the required energy is much larger than the required energy expected in the case of AWGN channel; that is, approximately 45 dB difference. This large amount of transmit power required for fading channel is caused by the nature of the fading channel, which has many deep nulls in its frequency response. If the signal is located at one of these nulls, the signal will then be attenuated considerably; thus, resulting in a very high BER. To reduce this high BER, more signal power is needed in order to compensate for such attenuation. Increasing signal power might not be a clever alternative especially in mobile applications since it then reduces the battery usable time. To overcome the problem from fading channel with an acceptable signal power, many techniques have been investigated; for example, channel encoding techniques, multicarrier modulation (especially Orthogonal Frequency Division Multiplexing (OFDM)), diversity coding (Kanprachar and Jacobs, 2003), and so on.

Spectral Efficiency

According to the ratio between the bit rate and the channel bandwidth, the spectral efficiency between these two standards can be compared. GSM has a bit rate of 270.833 kbps and a channel bandwidth of 200 kHz. USDC has a bit rate of 48.6 kbps and a channel bandwidth of 30 kHz. The spectral efficiency can be given by

$$\text{Spectral Efficiency} = \frac{\text{bit rate}}{\text{channel bandwidth}} \tag{24}$$

Using equation (24), it is found that USDC has a spectral efficiency of 1.62 bits/s/Hz, and GSM has a spectral efficiency of 1.35 bits/s/Hz. It means that USDC is better than GSM in terms of spectral efficiency. That is, USDC can send more information than GSM does, in the same channel bandwidth.

Conclusions

USDC and GSM are two standards for the digital wireless telephone system. USDC uses the channel bandwidth of 30 kHz with TDMA/FDD multiple access. Therefore, one channel bandwidth can support 3 users or up to 6 users for half-rate. The modulation type of USDC is a $\pi/4$ DQPSK with a raised cosine pulse shape, which has a rolloff factor of 0.35. This pulse shaping offers a very good performance to USDC in terms of zero ISI. However, this type of pulse shaping requires a linear amplification, which is not good in terms of power efficiency. Many detection techniques can be exploited in order to detect the signal. A coherent detection gives a better power efficiency than a differential detection does, with the same BER over a Gaussian channel. For a fading channel, a much better performance can be achieved by the coherent detection.

On the other hand, GSM uses the same multiple access but the channel bandwidth is 200 kHz supporting 8 users. GMSK modulation is used with Gaussian pulse shaping which has bandwidth-bit period product of 0.3. This type of pulse shaping makes this modulation non-linear; thus, a better power efficiency can be achieved by using a non-linear amplification. Coherent and non-coherent detection techniques can be used with this modulation type. Moreover, this pulse shaping offers small sidelobes with acceptable

Considering pulse shaping types, USDC is better than GSM in terms of zero ISI condition, which is achieved by using a raised cosine pulse shaping. Nevertheless, USDC requires a linear amplification, which has low power efficiency. In contrast, higher power efficiency amplification can be adopted in GSM. GMSK and $\pi/4$ DQPSK have the same characteristic of BER over a Gaussian channel. For a fading channel, GMSK and $\pi/4$ DQPSK will degrade significantly as the Doppler shift frequency increases (as seen by the BER floor in Figure 12). In terms of spectral efficiency, GSM with 1.35 b/s/Hz is worse than USDC, which has a spectral efficiency of 1.62 b/s/Hz.

In conclusion, the characteristics of USDC and GSM systems have been discussed and compared to one another. USDC system is better in terms of ISI problem and spectral efficiency while GSM system is better in terms of power efficiency. One important parameter in digital system is bit-error-rate (BER). It is seen that for both systems, BER are almost identical except for the case of AWGN channel in which the BER performance of USDC system is better than that of GSM system if coherent detection is adopted in USDC system.

References

Couch, L. (2001). *Digital and analog communication systems*.Upper Saddle River, NJ: Prentice-Hall.

Feher, K. (1991). MODEMS for emerging digital cellular-mobile radio system. *IEEE Transactions on Vehicular Technology,* 40, 355-365.

Goodman, D. (1991). Second generation wireless information networks. *IEEE Transactions on Vehicular Technology,* 40, 366-374.

Kanprachar, S., & Jacobs, I. (2003). Diversity coding for subcarrier multiplexing on multimode fibers. *IEEE Transactions on Communications*, 51 , 1546-1553.

Murota, K. (1981). GMSK modulation for digital radio telephony. *IEEE Transactions on Communications,* 29, 1044-1050.

Raith, K., & Uddenfeldt, J. (1991). Capacity of digital cellular TDMA systems. *IEEE Transactions on Vehicular Technology,* 40, 323-331.

Rappaport, T. (1996). *Wireless communications: Principles and Practice*. Upper Saddle River, NJ: Prentice-Hall.

Steele, R. (1992). *Mobile radio communications*. London: Pentech Press.

Yacoub, M. (1993). *Foundations of mobile radio engineering*. Boca Raton, FL: CRC Press, Inc.