A brief Overview of the GSM Radio Interface

Thierry Turletti

Telemedia Networks and Systems Group

Laboratory for Computer Science

Massachussets Institute of Technology

turletti@lcs.mit.edu

March 1, 1996

Abstract

This technical memorandum contains a compilation of several papers, reports and books relative to the GSM-900 radio interface. It is not exhaustive and it is restricted to the Traffic Channel/Full-Rate Speech (TCH/FS).

Keywords Base Station, Channel coding, FDMA, GMSK, GSM, Mobile Station, TDMA, Wireless Networking.

Contents

1 Introduction

The Global System for Mobile communications (GSM) is a digital cellular communications system initially developed in an European context which has rapidly gained acceptance and market share worldwide. It was designed to be compatible with ISDN systems and the services provided by GSM are a subset of the standard ISDN services (speech is the most basic).

The functional architecture of a GSM system can be divided into the Mobile Station (MS), the Base Station (BS), and the Network Subsystem (NS). The MS is carried by the subscriber, the BS subsystem controls the radio link with the MS and the NS performs the switching of calls between the mobile and other fixed or mobile network users as well as mobility management. The MS and the BS subsystem communicate across the Um interface also known as radio link.

Section 2 describes radio transmission aspects of GSM. Section 3 gives an overview of the channel coding operations.

2 Radio Transmission Aspects

For the GSM-900 system[1] , two frequency bands have been made available:

- 890 - 915 MHz for the uplink (direction MS to BS)
- 935 - 960 MHz for the downlink (direction BS to MS).

The 25 MHz bands are then divided into 124 pairs of frequency duplex channels with 200 kHz carrier spacing using Frequency Division Multiple Access (FDMA). Since it is not possible for a same cell to use two adjacent channels, the channel spacing can be said to be 200 kHz interleaved. One or more carrier frequencies are assigned to individual Base Station (BS) and a technique known as Time Division Multiple Access (TDMA) is used to split this 200 kHz radio channel into 8 time slots (which creates 8 logical channels). A logical channel is therefore defined by its frequency and the TDMA frame time slot number. By employing eight time slots, each channel transmits the digitized speech in a series of short bursts: a GSM terminal is only ever transmitting for one eighth of the time.

8-slot TDMA together with the 248 physical half-duplex channels corresponds to a total of 1984 logical half-duplex channels. This corresponds to roughly 283 (1984 / 7) logical half-duplex channels per cell. This is because a cell can only use one seventh of the total number of frequencies, see Figure 1. Seven sets of frequencies are sufficient to cover an arbitrarily large area, providing that the repeat-distance d is larger than twice the maximum radius r covered by each transmitter.

[1] Note that two alternative systems with additional capacity have been designed: the DCS-1800 and the PCS-1900 that operates respectively on 1.8GHz and 1.9 GHz carriers.

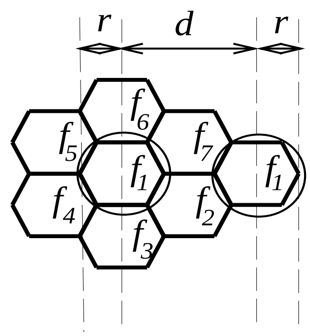

Figure 1: Typical cellular scheme

Each of the frequency channels is segmented into 8 time slots of length 0.577 ms (15/26 ms). The 8 time slots makes up a TDMA frame of length 4.615 ms (120/26 ms). The recurrence of one particular time slot every 4.615 ms makes up one basic channel.

The GSM system distinguishes between *traffic channels* (used for user data) and *control channels* (reserved for network management messages). In this overview, we consider only the Traffic Channel/Full-Rate Speech (TCH/FS) used to carry speech at 13 kbps.

TCHs for the uplink and downlink are separated in time by 3 burst periods, so that the mobile does not has to transmit and receive simultaneously. TCHs are defined using a 26-frame multiframe (i.e. a group of 26 TDMA frames). The length of a 26-frame multiframe is 120 ms, which is how the length of a burst period is defined (120 ms / 26 frames / 8 burst periods per frame). Out of the 26 frames, 24 are used for traffic, one is used for the Slow Associated Control Channel (SACCH) and one is currently unused (see Figure[2] 2).

Data are transmitted in bursts which are placed within the time slots. The transmission bit rate is 271 kb/s (bit period 3.79 microseconds). To allow for time alignment errors, time dispersion etc, the data burst is slightly shorter than the time slot (148 out of the 156.25 bit periods available within a time slot).

The burst is the transmission quantum of GSM. Its transmission takes place during a time window lasting (576 + 12/13) microseconds, i.e. (156 + 1/4) bit duration. A normal burst contains two packets of 58 bits (57 data bits + 1 stealing bit) surrounding a training sequence of 26 bits. The 26-bit training sequence is of a known pattern that is compared with the received pattern in order to reconstruct the rest of the original signal (multipath equalization). The actual implementation of the equalizer is not specified in the GSM specifications. Three "tail" bits are added on each side.

GSM can use slow frequency hopping where the mobile station and the base station transmit each TDMA frame on a different carrier frequency. The frequency hopping algorithm is broadcast on the Broadcast Control Channel. Since multipath fading is dependent on carrier frequency, slow frequency hopping help mitigate the problem. Fre-

[2]This figure is inspired by [4].

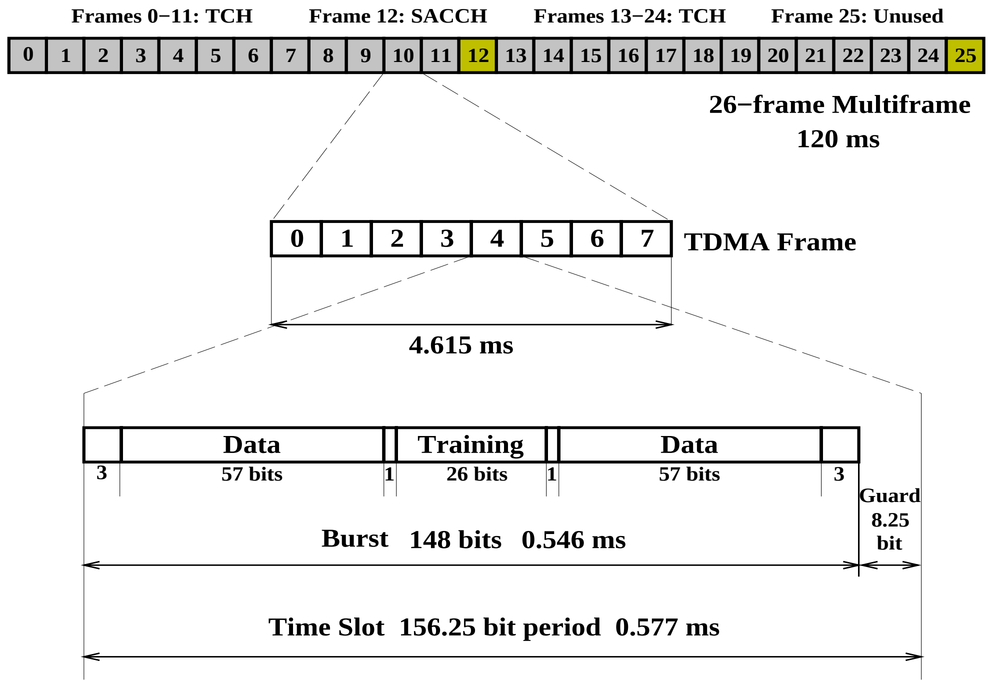

Figure 2: The TDMA frame structure

quency hopping is an option for each individual cell and a base station is not required to support this feature.

3 From Speech to Radio Waves

Figure 3 depicted the sequence of operations from speech to radio waves and from radio waves to speech. These operations are described in the following sections.

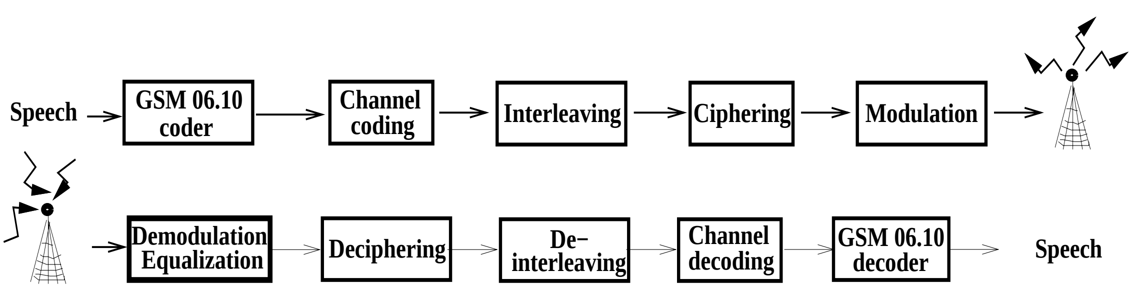

Figure 3: The sequence of operations

3.1 The GSM Speech Coding

The full rate speech codec in GSM is described as Regular Pulse Excitation with Long Term Prediction (GSM 06.10 RPE-LTP). A good overview of this algorithm has been done by Jutta Degener and Carsten Bormann at the Technical University of Berlin[3]. Moreover, they have developed a software implementation of the GSM 06.10 speech codec which is available in the public domain. Basically, the encoder divides the speech into short-term predictable parts, long-term predictable part and the remaining residual pulse. Then, it encodes that pulse and parameters for the two predictors. The decoder reconstructs the speech by passing the residual pulse first through the long-term prediction filter, and then through the short-term predictor, see Figure 4. Note that the

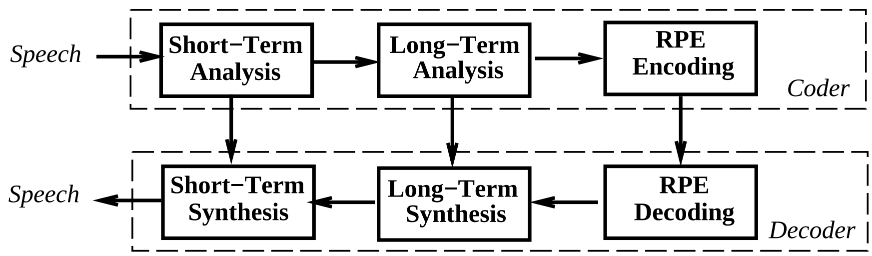

Figure 4: A block diagram of the GSM 06.10 codec

Phase 2 of GSM defines a new half rate speech encoder (GSM 06.20 RPE-LTP).

3.2 The GSM Channel Coding

Channel coding introduces redundancy into the data flow in order to allow the detection or even the correction of bit errors introduced during the transmission [10].

The speech coding algorithm produces a speech block of 260 bits every 20 ms (i.e. bit rate 13 kbit/s). In the decoder, these speech blocks are decoded and converted to 13 bit uniformly coded speech samples. The 260 bits of the speech block are classified into two groups. The 78 Class II bits are considered of less importance and are unprotected. The 182 Class I bits are split into 50 Class Ia bits and 132 Class Ib bits (See Figure 5). Class Ia bits are first protected by 3 parity bits for error detection. Class Ib bits are then added together with 4 tail bits before applying the convolutional code with rate $r = \frac{1}{2}$ and constraint length $K = 5$. The resulting 378 bits are then added to the 78 unprotected Class II bits resulting in a complete coded speech frame of 456 bits (see Figure 6).

[3]See URL: "http://www.cs.tu-berlin.de/ jutta/toast.html"

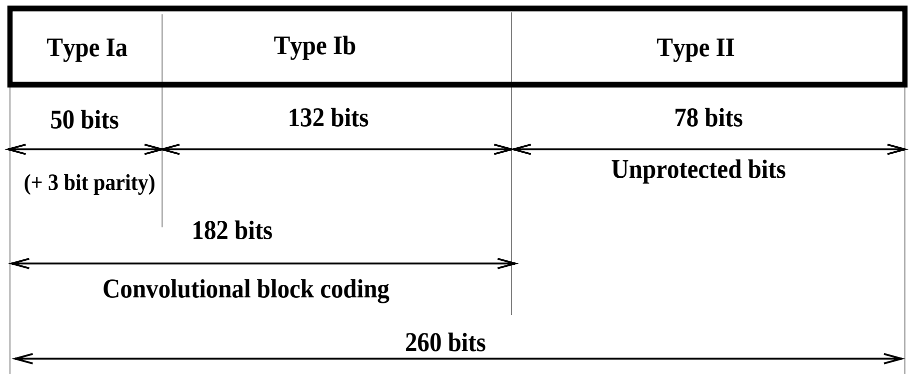

Figure 5: Audio sample: 1 block = 260 bits (20 ms)

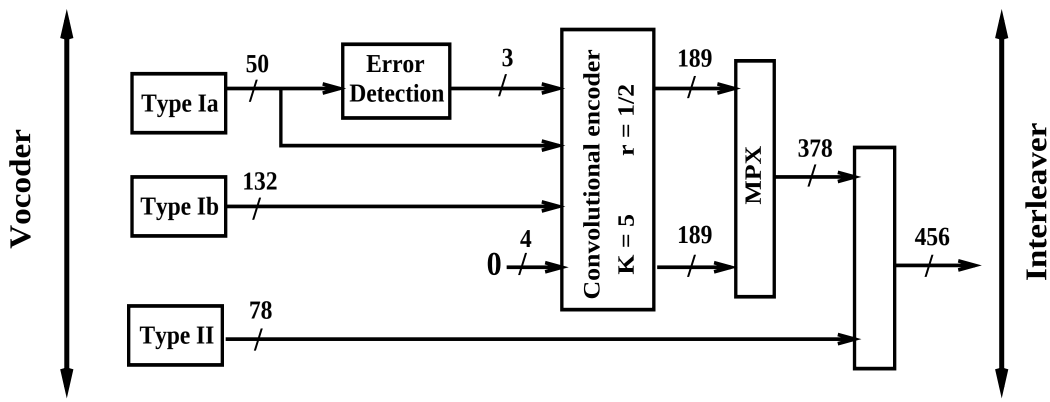

Figure 6: TCH/FS Transmission Mode

3.2.1 Error Detecting Codes

The GSM standard uses a 3-bit error redundancy code to enable assessment of the correctness of the bits which are more sensitive to errors in the speech frame (the category Ia 50-bits). If one of these bits are wrong, this may create a loud noise instead of the 20 ms speech slice. Detecting such errors allows the corrupted block to be replaced by something less disturbing (such as an extrapolation of the preceding block).

The polynomial representing the detection code for category Ia bits is $G(X) = X^3 + X + 1$.

At the receiving side, the same operation is done and if the remainder differs, an error is detected and the audio frame is eventually discarded.

3.2.2 Convolutional Coding / Decoding

Convolutional coding consists in transmitting the results of convolutions of the source sequence using different convolution formulas. The GSM convolutional code consists in adding 4 bits (set to "0") to the initial 185 bit sequence and then applying two different convolutions: polynomials are respectively $G_1(X) = X^4 + X^3 + 1$ and $G_2(X) = X^4 + X^3 + X + 1$. The final result is composed of twice 189 bits sequences, see Figure 6.

Convolutional decoding can be performed using a Viterbi algorithm [2]. A Viterbi decoder logically explores in parallel every possible user data in sequence. It encodes and compare each one against the received sequence and picks up the closest match: it is a maximum likelihood decoder. To reduce the complexity (the number of possible data sequence double with each additional data bit), the decoder recognizes at each point that certain sequences cannot belong to the maximum likelihood path and it discards them. The encoder memory is limited to K bits; a Viterbi decoder in steady-state operation keeps only 2^{K-1} paths. Its complexity increases exponentially with the constraint length K.

The GSM convolutional coding rate per data flow is 378 bits each 20 ms, i.e.: 18.9 kb/s. However, before modulate this signal, the 78 unprotected Class II bits are added (see Figure 6.). So, the GSM bit rate per flow is 456 bits each 20 ms i.e. 22.8 kb/s.

Note that there is a software Viterbi decoder developed [4] by Phil Karn, from Qualcomm Inc. which supports the (K=7, r=1/2) NASA standard code [3].

3.3 Interleaving / De-interleaving

Interleaving is meant to decorrelate the relative positions of the bits respectively in the code words and in the modulated radio bursts. The aim of the interleaving algorithm is to avoid the risk of loosing consecutive data bits. GSM blocks of full rate speech are interleaved on 8 bursts: the 456 bits of one block are split into 8 bursts in sub-blocks of 57 bits each. A sub-block is defined as either the odd- or the even-numbered bits of the coded data within one burst. Each sub-blocks of 57 bit is carried by a different burst and in a different TDMA frame. So, a burst contains the contribution of two successive speech blocks A and B. In order to destroy the proximity relations between successive bits, bits of block A use the even positions inside the burst and bits of block B, the odd positions (see Figure 7).

De-interleaving consists in performing the reverse operation. The major drawback of interleaving is the corresponding delay: transmission time from the first burst to the last one in a block is equal to 8 TDMA frames (i.e. about 37 ms).

3.4 Ciphering / Deciphering

A protection has been introduced in GSM by means of transmission ciphering. The ciphering method does not depend on the type of data to be transmitted (speech, user data or signaling) but is only applied to normal bursts.

Ciphering is achieved by performing an "exclusive or" operation between a pseudo-random bit sequence and 114 useful bits of a normal burst (i.e. all information bits

[4] See URL "http://www.qualcomm.com/people/pkarn/ham.html".

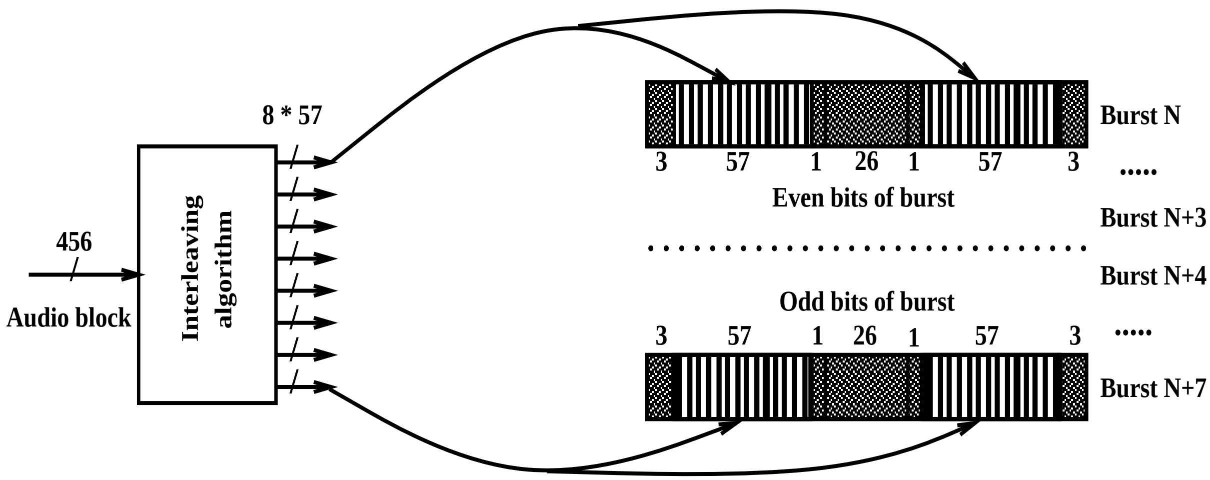

Figure 7: Interleaving operation

except the 2 stealing flags). The pseudo-random sequence is derived from the burst number and a key session established previously through signaling means. Deciphering follows exactly the same operation.

3.5 Modulation / Demodulation

GSM uses the Gaussian Modulation Shift Keying (GMSK) with modulation index $h = 0.5$, BT (filter bandwidth times bit period) equal to 0.3 and a modulation rate of 271 (270 5/6) kbauds. The GMSK modulation has been chosen as a compromise between a fairly high spectrum efficiency (of the order of 1 bit/Hz) and a reasonable demodulation complexity. The constant envelope allows the use of simple power amplifiers and the low out-of-band radiation minimizes the effect of adjacent channel interference. GMSK differs from Minimum Shift Keying (MSK) in that a pre-modulation Gaussian filter is used. The time-domain impulse response of the filter is described in Equation 1, where $k_1 = \frac{\pi}{\sqrt{2 \ln 2}}$ and B is the half-power bandwidth.

$$h(t) = \frac{k_1 B}{\sqrt{\pi}} e^{-k_1^2 B^2 t^2} \tag{1}$$

A block diagram of a GMSK modulator in Figure 8.

The Viterbi algorithm can also be used as a Maximum Likelihood Sequence Estimator (MLSE) equalizer [1, 2, 8]. So a GSM receiver can contain two different implementations of the Viterbi algorithm.

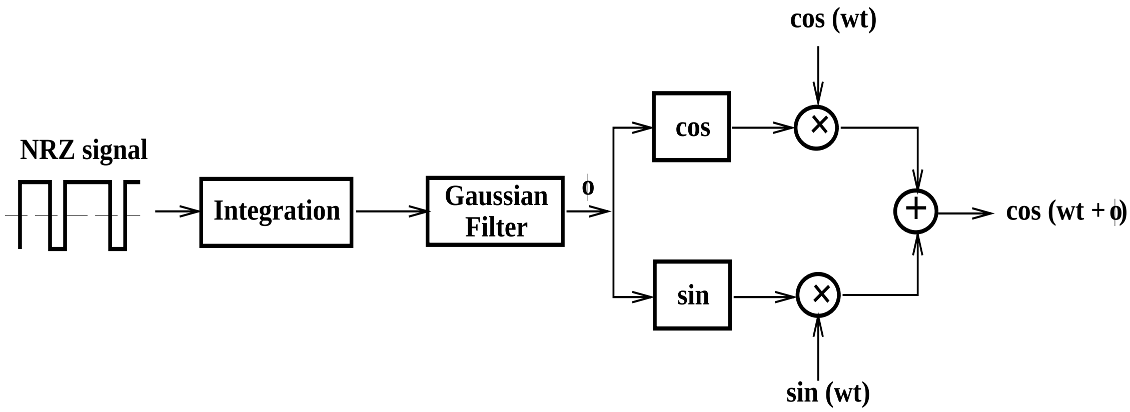

Figure 8: GMSK modulation block diagram

Power Class	Maximum Power of a Mobile Station / (dBm)	Maximum Power of a Base Station / (dBm)
1	20 W (43)	320 W (55)
2	8 W (39)	160 W (52)
3	5 W (37)	80 W (49)
4	2 W (33)	40 W (46)
5	0.8 W (29)	20 W (43)
6		10 W (40)
7		5 W (37)
8		2.5 W (34)

Table 1: Power Levels in the GSM System

3.6 RF Power levels

Radio equipment in GSM can be classified by the various power classes that correspond to different transmitter power levels. Table 1 shows[5] the characteristics of each power class for both mobile stations and base stations. The minimum mobile station power level is 20 mW (13 dBm).

Acknowledgements

This work was supported by the Advanced Research Projects Agency under contract No. F30602-92-C-0019 (monitored by AFSC, Rome Laboratory) and contract No. DABT-6395-C-0060 (monitored by US Army, Fort Huachuca).

[5] This Table has been reproduced from [9].

References

[1] Baier, A., Heinrich, G. and Wellens, U. "Bit synchronization and timing sensitivity in Adaptive Viterbi Equalizers for narrowband-TDMA digital mobile radio systems", *Proc. of the 38th IEEE Vehicular Technology Conference*, 1988, pp. 377-384.

[2] Forney, G.D. JR. "The Viterbi Algorithm" *Proc. of the IEEE*, Vol. 61, No 3, Mar. 1973, pp. 268-278.

[3] Karn, P. "Convolutional decoders for amateur packet radio", *to appear at the 1995 ARRL Digital Communications Conference.*

[4] Lycksell, E. "GSM System overview", *Swedish Telecom Radio Internal Report*, Jan. 1991.

[5] Margrave, D. "Computer Simulation of the Radio Channel Aspects of the GSM System", *ECE 798 Research Project*, see URL "http://www.utw.com/ dmargrav/gsm/ece798/paper.html", Nov. 1995.

[6] Murota, K. and Hirade, K. "GMSK Modulation for Digital Radio Telephony", *IEEE Trans. on Communications*, Vol. com-29, No 7, July 1981, pp. 1044-1050.

[7] Mouly, M. and Pautet, M-B. "The GSM System for Mobile", *ISBN: 2-9507190-0-7*, 1992.

[8] Ono, S., Hayashi, H., Tanak, T. and Kondoh, N. "A MLSE receiver for the GSM digital cellular system", *Proc. of the 44th IEEE Vehicular Technology Conference*, June 8-10, 1994, pp. 230-233.

[9] Redl, S. M., Weber, M. K. and Oliphant, M. W. "An Introduction to GSM" *ISBN: 0-89006-785-6*, 1995, British Library Cataloguing in Publication Data.

[10] Wakerly, J. "Error detecting codes, self-checking circuits and applications", *Computer design and architecture series, the computer science library*, Edward J. McCluskey Editor, 1978.